

1978

An instrumentation system for automatically monitoring and recording sound levels in industrial arts laboratories

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AN INSTRUMENTATION SYSTEM FOR AUTOMATICALLY
MONITORING AND RECORDING SOUND LEVELS IN
INDUSTRIAL ARTS LABORATORIES.

IOWA STATE UNIVERSITY, PH.D., 1978

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An instrumentation system for automatically monitoring
and recording sound levels in industrial arts laboratories

by

W. Wayne Weber

A Dissertation Submitted to the
Graduate Faculty in Partial Fulfillment of
The Requirements for the Degree of
DOCTOR OF PHILOSOPHY

Major: Industrial Education

Approved:

Signature was redacted for privacy.

In Charge of Major Work

Signature was redacted for privacy.

For the Major Department

Signature was redacted for privacy.

For ~~the~~ Graduate College

Iowa State University
Ames, Iowa

1978

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ABSTRACT

The problem of the study was to fabricate and test the sensitivity and repeatability of a sound monitoring system through the use of variable noise levels produced by a variety of machines.

Specifically, the objective of the study was to design and fabricate a sound monitoring system capable of sensing and recording for computer processing of data, multi-locational sound levels over extended periods of time. In regard to the objective the following questions were presented.

(1) Is the sound monitoring system responsive to locational changes of variable noise levels emanating from identified sources?

(2) Is the sound monitoring system capable of providing repeated recordings of sound levels within acceptable tolerances?

(3) Is the system adaptable for detecting and recording noise levels in typical industrial arts laboratories?

The instrumentation system consisted of an approved type 2 sound level meter (SLM) which was used in conjunction with a 16 microphone matrix. An electronic switcher-converter was fabricated to serially switch each microphone of the matrix at precise timed intervals with each matrix cycle being registered by a digital counter. The linear voltage output of the SLM was converted to digital form and recorded on punched tape for computer processing. A punch tape unit designed for the system was used to perform the recording function.

The instrumentation system was tested in the wood laboratory of the Industrial Education Department at Iowa State University, where the microphones were arranged in a four by four matrix spaced at 10 foot intervals.

The microphones were switched at four second intervals (64 seconds for one matrix cycle). Various machines and combinations of machines made up the 23 sound treatments, each of which were monitored for a period of four matrix cycles.

The data on the punched tape were converted by computer processing to decibel form and read out in a row-column format, with the columns representing the microphone positions and the rows representing the treatments. The four matrix cycle average per treatment for each microphone position was used as a basis for computer development of sound contour maps for each treatment.

It was found that the system was responsive to locational changes of variable sound sources, that it was capable of providing repeated readings within acceptable tolerances, and that it was adaptable for detecting and recording noise levels in typical industrial arts laboratories.

CHAPTER I. INTRODUCTION

Man's responses to the sounds of today are based on biological conditioning derived from the environment of the distant past. In this regard an important contrast can be drawn between the senses of sound and sight.

Since man's early beginning, the passing of each day has brought about great changes in the intensity of light on earth. These changes are on the order of ten billion to one when comparing the darkest night with the brightest day. Also, the everyday processes of survival, the moving from dark forests and caves to bright plains, created a need for sight receptors with a capacity for great changes in sensitivity. Nature responded with the provision of eye lids to exclude light and pupils that varied in size with changes in light intensity.

The environment of the past for sound and hearing offered a different set of adaptive circumstances. As the ear developed, it did not meet with the great daily variations in sound levels. While it had to withstand the intense but brief sounds of thunder and the moderate sounds of wind and rain, these rarely lasted more than a few hours at infrequent intervals (Miller, 1974, p. 730).

Man's hearing evolved in the relative quiet of a past environment, and became a dominant factor in his adaptation to the environment on an individual and social basis. For the individual, sound through the sense of hearing serves as a constant sentinel, always on the alert for danger. It provides input to man's arousal and activation systems and stimulates certain muscular reflexes to cause the head and eyes to turn in the

direction of a sound source to aid in recognition and identification. Socially man uses his voice and his ears to produce and communicate his past experiences and present ideas and intentions by means of language. Man's love of music and sense of rhythm are associated with hearing and are culturally and socially determined along with the effects of sound on man's temperament and feeling of well-being.

Man evolved in a world of sound, and as a result of this adaptive process, sound is of great value to man. But not all sound is useful or desirable. Some is classified as noise. Warring (1969) stated:

The definition of noise as being unwanted sound cannot be bettered. The definition introduces the concept of the inter-relation between, on the one hand, human beings (and perhaps animals too) and their idiosyncrasies and environmental requirements and on the other, the physical stimulus of sound. (p. 7)

The decision to categorize a particular sound as noise requires a value judgment. It calls for an agreement among the members of a society or group. Baron (1970) commented about how sound becomes noise:

There is no simple answer to this question, any more than there is a simple answer to the question of what is an optimum acoustic environment. We can only hope to approximate an answer.

In general, sound is noise when its physical components disturb the relationship between man and his fellow man, and man and his environment. Or when the acoustic energy causes undue stress and actual physiological damage.

In conventional terms, sound may be classified as noise when it damages the hearing mechanism, causes other bodily effects detrimental to health and safety, disturbs sleep and rest, interferes with conversation or other forms of communication, annoys or irritates. (p. 46)

Prehistoric man, whose hearing evolved in a world of relative quiet where sound conveyed information vital for survival, would not have experienced noise in terms of the previous definition. Instead it was

during a comparatively small time span from the dawn of recorded history to the present that sound became noise. Baron (1970) stated:

Noise is nothing new. Even the main reasons for most noise--convenience and speed--are as old as the first squeaky wheel. Chariots racing noisily over Rome's cobblestones forced Julius Ceasar to try--unsuccessfully--to ban daytime charioteering.

Some would like us to think we have merely traded new noises for old, but the acoustic attack on man and his environment really began in earnest with the Industrial Revolution. From a predominantly agricultural husbandry, man found himself uprooted by the pull of the factories to the grime and congestion of the cities. He found himself surrounded on all sides by factories making millions of devices to enable him to speed over the surface of the earth, soar through the skies, and blend orange juice. (p. 22)

The questions can now be asked. What kind of a noise environment has modern technology created? What noise level is judged to be excessive?

The sounds and noise common to everyday living can be measured in several ways. A convenient method is to represent sound in terms of sound pressure levels on the A-weighted decibel scale (dBA). The decibel is a nonlinear unit based on powers of 10 to express the ratio between a given level and some arbitrary reference level. For example, 10 decibels is 10 times more intense than one decibel, 20 decibels is 100 times (10×10) more intense, 30 decibels is 1000 times ($10 \times 10 \times 10$) more intense, and continuing to where 100 decibels is 10 billion times as great as one decibel. This scale is used because the human ear can detect such a wide range of sound energy. Peterson & Gross (1974) list the relative levels of many common sounds and noise, as shown in Table 1. It can be noted that sound levels emanating from many of the machines used for work and recreation create sound levels that range high on the

Table 1. Typical A-weighted sound levels (Peterson & Gross, 1974, p. 4)

At a given distance from noise source	Decibels ₂ Re 20 N n ²	Environmental
	140	
50 HP SIREN (100')	130	
JET TAKEOFF (200')	120	
RIVETING MACHINE	110	CASTING SHAKEOUT AREA
CUT OFF SAW PNEUMATIC PEEN HAMMER	100	ELECTRIC FURNACE AREA
TEXTILE WEAVING PLANT SUBWAY TRAIN (20')	90	BOILER ROOM PRINTING PRESS PLANT
PNEUMATIC DRILL (50')	80	TABULATING ROOM INSIDE SPORT CAR (50 MPH)
FREIGHT TRAIN (100') VACUUM CLEANER (10') SPEECH (1')	70	
	60	NEAR FREEWAY (auto traffic) LARGE STORE ACCOUNTING OFFICE
LARGE TRANSFORMER (200')	50	PRIVATE BUSINESS OFFICE LIGHT TRAFFIC (100') AVERAGE RESIDENCE
	40	MIN LEVELS RESIDENTIAL AREAS IN CHICAGO AT NIGHT
SOFT WHISPER (5')	30	STUDIO (speech)
	20	STUDIO FOR SOUND PICTURES
	10	
THRESHOLD OF HEARING YOUTHS 1000-4000 H _z	0	

dBA scale; however, the determination of harmful sound levels as stipulated by current federal and state regulations are presented in terms of both sound level and exposure time.

The maximum allowable noise exposure levels in a working environment were set forth in the Occupational Safety and Health Act (OSHA) enacted in 1971, and have been included in the Iowa Occupational Health and Safety Act enacted by the Sixty-fifth General Assembly, effective July 1, 1974. These standards are listed in Table 2 as presented in the Federal Register (1974).

Table 2. Permissible noise exposures^a (Federal Register, 1974, p. 23597)

Duration per day, hours		Sound level dBA slow response
8	90
6	92
4	95
3	97
2	100
1 1/2	102
1	105
1/2	110
1/4 or less	115

^aWhen the daily noise exposure is composed of two or more periods of noise exposure of different levels, their combined effect should be considered, rather than the individual effect of each. If the sum of the following fractions: $C_1/T_1 + C_2/T_2 + \dots + C_n/T_n$ exceeds unity, then, the mixed exposure should be considered to exceed the limit value. C_n indicates the total time of exposure at a specified noise level, and T_n indicates the total time of exposure permitted at that level.

Exposure to impulsive or impact noise should not exceed 140 dB peak sound pressure level.

These OSHA standards are a natural result of the concern for the increasing sound or noise levels accompanying the speed and convenience

of modern technology. While OSHA regulations on the federal level were originally intended for the industrial sector, by state law these regulations also apply to school facilities. Industrial education is an educational discipline that teaches about industry and technology, and in the process, laboratory activities involving machines and processes of industry are used. The main endeavor of this research project was to design, fabricate, and test an instrumentation system that would monitor and record for computer processing the sound environment that exists in industrial arts laboratories.

Problem of the Study

The problem of the study was to fabricate and test the sensitivity and repeatability of a sound monitoring system through the use of variable noise levels produced by a variety of machines.

Purpose of the Study

The purpose of this study was twofold:

1. To develop sound monitoring instrumentation to sense and record for computer processing at predetermined intervals, the sound levels present at varied locations in industrial arts laboratories.
2. To develop instrumentation capabilities to sense and record at precise timed intervals for extended periods of time, the noise levels in industrial arts laboratories to provide a time-oriented sound profile for the determination of compliance with OSHA regulations.

Need for the Study

The need for the study was established by the documentation of information from the literature relative to noise and hearing. The topic was examined from the following perspectives: (a) the occupational and environmental aspects of noise pollution, (b) measurement of sound, (c) noise in the public school, and (d) the legal ramifications pertaining to schools.

Occupational and environmental aspects

In Introduction to Sound and Hearing, Davis (1965) stated:

We attach arbitrary and abstract meanings to sounds, and we have language. We communicate our experiences of the past and also our ideas and plans for the future action. For human beings, then, the loss of hearing brings special problems and a special tragedy....human society creates a special problem even for those with perfect hearing--the problem of unwanted sound, of noise, which is as much a hazard of our environment as disease or air pollution. (p. 2)

While commenting on the comparison between the man-made noises of today's technical society and the sounds of the past, Ward (1970a) offered a historical observation to the effect that ancient travelers were aware that villagers who lived close to the cataracts of the Nile appeared to be hard of hearing (p. 22). Although Ward did not indicate whether the people living during those times were aware of any relationship between the water noise and loss of hearing, these observations are significant in terms of modern-day sound studies.

In their study relative to hearing loss due to noise exposure and aging, Gallo and Glorig (1964) stated that with as few as 10 years of exposure to high-level (90 dB sound pressure level) industrial plant

noise, men thirty years of age may experience hearing level loss in excess of that in men twice their age who have been subjected to a low noise environment.

The findings of a thorough study of the hearing loss experienced by female workers in the jute weaving mills of Scotland (Taylor, Pearson, Mair, & Burns, 1965) reported that the 96 dBA sound level environment of these mills will cause the hearing threshold of a young worker to shift temporarily 35-65 dB in the 4000 Hz range the first day on the job. It must be stressed that this is a most critical frequency and the shift would not be permanent from one day's exposure, but after 10 to 15 years on the job, the workers in the jute mills suffered extensive hearing losses as compared to individuals in the same geographical location who were not exposed to the sound environment of the jute mills.

Commenting on this study, Miller (1974) stated:

Evidently, as the exposures are repeated year after year, the ear becomes less and less able to recover from the temporary threshold shift present at the end of each day.... (p. 736)

Miller further stated that since there are only 16 hours of time for the ears of these workers to recover between work-day exposures, most of the workers will be living with the chronic threshold shift of 25-55 dB at 4000 Hz during the week even from the first day of employment. Because of the greater recovery time, younger workers will have nearly normal hearing on Saturday and Sunday, but as the years pass these jute weavers will become partially deaf.

Similar data were collected on male workers in noisy industries in the United States (Nixon & Glorig, 1961). This study reported that

workers with 10 years exposure at 83 dB have a hearing threshold shift of 10-12 dB. At an exposure level of 92 dB, the shift is 25 dB and at an exposure level of 97 dB, the shift is 45 dB.

Measurement of sound

A sound measurement study conducted by Pinder (1974) of woodworking machines used in a New Jersey school system indicated that certain common power machines created excessive noise levels. For example, the Rockwell/Delta uniplane created a noise level of 115 dB. Other measurements included the Oliver tilting arbor table saw at 105 dB, the Parks 20-inch planer at 100 dB, the Rockwell/Delta heavy duty shaper at 95 dB, and the Yates American 16-inch jointer at 90 dB.

It should be noted that these noise levels are high when compared to occupational OSHA standards; however, Pinder compared his findings with the more stringent nonoccupational OSHA standards which, for example, allow exposure to 115 dB levels for a maximum of two minutes as compared to the occupational standards of 15 minutes. By way of comparison, Pinder offered this comment:

The OSHA occupational limits are perhaps viewed as acceptable in industry because the worker receives a paycheck for assuming certain job-related risks. For these reasons, the lower limits (non-occupational) were adopted as a standard which would prevent any notable losses for virtually everyone exposed in a teaching-learning environment.

It is likely that during a typical class period in woodworking, the non-occupational noise intensity and duration limits will be exceeded. Moreover, the students may be subjected to possible harmful noise conditions which exceed OSHA occupational limits.
(p. 48)

Noise and the public school

The luxuries of speed, convenience, and improved living standards enjoyed in the developed countries as the result of their advanced technology, have, in the long term, created problems of environmental destruction and pollution that have built up to serious proportions in the occupational, recreational, and family activities. In the United States, the public schools, as an inseparable part of our society, must not be considered as being immune to these effects of technology.

As the result of a sound level study conducted in schools in eastern Texas, Turner (1970) stated:

The type activity in the classroom made a highly significant difference in the classroom sound level in both the elementary and secondary schools. Noise levels above 85 dB can produce hearing loss depending upon the length and frequency of exposure and the intensity and nature of the noise. Since readings of 90 dB and above were found in all secondary band rooms and in some shop and P.E. classes, it would appear that noise levels are high enough to cause concern as to the possibility of hearing loss for some students. (p. 12)

Commenting in regard to technology and our environment, John Feirer, executive editor, Industrial Arts and Vocational Education, stated:

Since industrial education is a study of industry and technology, we must certainly be concerned with its effect on environment. This cannot be a theoretical, "hold your breath" concern. If industrial education is to play an active role in this area, every teacher must be concerned about how he can relate his technical area to environment. What better place to start than in the lab itself where students spend an hour or two a day and where teachers live their working hours? Many of these shops are "micro-environments" in themselves. Wherever we have dust, heat, dirt and noise, we have environmental problems....What kind of sound conditioning do we have in the metals lab? How are combustion fumes exhausted in the power and auto labs?

These are practical everyday applications of environmental control. They should be seriously studied by students and teachers as a part of their industrial education program (Feirer, 1971, p. 21).

In a more recent editorial in Industrial Education (formerly Industrial Arts and Vocational Education), Feirer (1973) commented on safety standards and federal legislation:

The current federal legislation, namely the Occupational Safety and Health Act and the Construction Safety Act of 1970 have added new dimensions to safety standards. These Acts which became law in August, 1971, established federal safety standards for industry. Most of these very same standards apply to school shops and laboratories. Many newer safety regulations go far beyond such things as safety glasses and guards for machines. On the matter of noise, workers may not be exposed to a sound level of 97 decibels longer than three hours a day. Most school shop wood labs having a planer would greatly exceed this sound level. (p. 23)

As a part of a noise pollution study of Utah public schools, Hicks (1974) measured the sound levels in 18 industrial arts wood laboratories and 12 metals laboratories. The findings indicated that the noisiest of the woodworking machines was the surface planer with a noise output of 107 dB. The radial arm saw was second at 100 dB, followed by the table saw at 97.5 dB. In regard to the metals laboratories, Hicks stated:

The loudest three pieces of equipment found in the metals laboratories were: (1) the foundry furnace at an average of 102.6 decibels; (2) the pedestal grinder at 97.1, and (3) the arc welder at 82.3 decibels. (p. 120)

As a summary of his study, Hicks concluded that as a result of the noise pollution in the industrial arts laboratories, a potential safety and health hazard exists. He stated that while more laboratories are in compliance with OSHA regulations, many do exceed the allowable exposure considered to be harmful to health and safety. Finally he stressed that the sound levels that exist around machinery must not be considered as a problem confined to industry because industry machinery used in an

educational setting produces the same potential for the impairment of hearing.

Legal ramifications

The culmination of man's growing awareness and concern for the effects of his technology on his environment and thus on his ultimate health and survival was manifested by the passage of the Occupational Safety and Health Act (OSHA) of 1970. This federal legislation represented the beginning of enforceable standards for the working environment in this country. Subsequently, the standards of this legislation are used by many states, including Iowa, as the basis for state OSHA laws. As set forth in the Code of Iowa (1975), the acts of the Sixty-fifth General Assembly, effective July 1, 1974, applied to schools as follows:

Every student and teacher in any public or nonpublic school shall wear industrial quality ear-protective devices while the student or teacher is participating in any phase or activity of a course which may subject the student or teacher to the risk or hazard of hearing loss from noise in processes or procedures used in any of the following courses:

1. Vocational or industrial arts shops or laboratories involving experiences with any of the following:
 - a. Milling, sawing, turning, shaping, cutting, grinding, or stamping of any solid materials
 - b. Kiln firing of any metal or other materials
 - c. Electric arc welding
 - d. Repair or servicing of any vehicles while in shop
 - e. Static tests, maintenance or repair of internal combustion engines
 - f. Letter press, paper folders, monotype.

"Noise" as used in this section, means a noise level that meets or exceeds damage-risk criteria established by the present federal standards for occupational noise exposure, Occupational Safety and Health Standards. (p. 253)

Summary of the need for the study

Down through the ages and up to and including the early years after the Industrial Revolution, man could justifiably ignore the possibility that excessive environmental noise could present a hazard for man. The justification for this oversight was based on innocence due to the lack of knowledge about the effects of noise on man. Today this claim of innocence is not a tenable argument. The findings of the studies cited reported that hearing loss is caused by excessive exposure to noise in common occupational settings in this age of technology. Also, sound level studies made in school laboratories with conventional manually-operated sound level meters have shown that hazardous sound levels existed during certain measurement periods. Because of the growing concern for these hazards, the Iowa Occupational Health and Safety laws, IOSHA, as amended by the Sixty-fifth General Assembly, stated that all schools in Iowa are to comply with the current federal OSHA standards for noise exposure.

In order for administrators to make valid decisions concerning compliance with this law, accurate sound level measurements of a more extensive nature than practical by manual methods was deemed necessary. The instrumentation developed for this study does accurately sense and record sound levels at multiple laboratory locations with measurements made at precise intervals during a part of, or for an entire activity day.

Questions of the Study

The objective of the study was to design and fabricate a sound monitoring system capable of sensing and recording for computer processing of data, multi-locational sound levels over extended periods of time. In accordance with the objective, the following questions were presented.

- (1) Is the sound monitoring system responsive to locational changes of variable noise levels emanating from identified sources?
- (2) Is the sound monitoring system capable of providing repeated recordings of sound levels within acceptable tolerances?
- (3) Is the system adaptable for detecting and recording noise levels in typical industrial arts laboratories?

Assumptions of the Study

- (1) The monitoring system could be technically constructed with inherent sensitivity and repeatability capabilities.
- (2) The sound standards used for testing were valid representations of typical noise sources in industrial arts laboratories.

Delimitations of the Study

- (1) The study was limited by the accuracy of The American National Standards Institute (ANSI) approved Type 2 sound level meter used as a measurement standard.
- (2) The sound level readings were limited to those made by A-weighted slow response settings on the sound level meter.
- (3) The sound level readings were limited by the response of a matrix of microphones mounted in fixed matrix positions.

Procedure of the Study

The procedure of this study consisted of four parts: (1) design and fabrication of the instrumentation, (2) testing and initial calibration of the instrumentation, (3) collection of data, and (4) analysis of data.

Design and fabrication of the instrumentation

(a) An approved sound level meter was selected with provision for detaching the microphone for remote use up to distances of 50 feet.

(b) Sixteen microphones and associated extension cables were secured for purposes of this experiment.

(c) A solid state electronic switcher-converter was designed and constructed.

1) The device was designed to select microphones and switch them in sequence at accurately-timed intervals

2) A second function of this device was to convert analog signals to digital form so that linear relationship existed between input voltage and output signal counts.

(d) An eight-level punch tape unit was designed and constructed to record in binary format the output counts from the converter.

Testing and initial calibration of the instrumentation

First in the calibration process of the instrumentation, it was necessary to:

(a) determine the half- and full-scale DC voltages across the input to the meter movement of the sound level meter

(b) set the gain of the linear analog amplifier into the analog to digital converter so that an output of two and one-half volts was produced for half scale on the sound level meter and five volts for full scale

(c) calibrate the analog to digital converter so that an output count of 100 resulted from two and one-half volt input and an input of five volts produced in a linear manner a count of 200 with provision for an overcount up to 250.

Collection of data

(a) Arranged the 16 microphones to form a four-by-four matrix equally spaced within the selected wood laboratory.

(b) Compensated for variations in microphones by adjusting the calibrator for each microphone position.

(c) Set up a log listing the machine or combination of machines for each treatment situation.

(d) Recorded each treatment sound level for a period of four complete matrix cycles (64 readings).

Each reading was associated with a certain microphone and occupied a timed interval which included a subsequence of four counts. On the first count the system switched to the next microphone in the 16-microphone matrix sequence and sound monitoring from this microphone did begin. Also, the digital data from the previous microphone position was recorded in binary form on punched tape. During the second count, sound monitoring continued and a logic test was conducted to determine if the system was switched to the number one microphone of the matrix. If this

was true, an end of matrix sequence punch was executed to aid the computer retrieval of the data. The monitoring process continued during the third count, and at the beginning of the fourth count, the sound level was converted to digital form and stored as a binary number to be read out and recorded on the number one count of the next reading sequence.

The format of the punched tape was straightforward. The tape was punched in eight tracks or binary levels that related to powers of two, ranging from 2^0 to 2^7 (decimal 1 to 255). Beginning with the sound level information monitored by microphone number one, the tape was punched in sequence with microphone number two following number one and continuing around the entire matrix to microphone 16. For a given punch, a hole in a given track indicated that a power of two of that level was part of the total number for that punched recording. After each recording for microphone 16, an end-of-sequence punch, which consisted of a punch or hole for each level (binary 11111111), was executed. This punch enabled the computer to begin a new count for each matrix cycle and eliminated the progressive microphone identification errors that could develop if the computer was programmed to count from the first punch.

The data in binary form was read from the punched tape into the computer where the binary count was changed to decimal form and stored in a row-column format. There was a column for each of the 16 microphones and a row for each matrix cycle. This input data was changed to decibel form for each reading and printed in row-column format. In addition, the average reading at each microphone position for each treatment

appeared in the printout and was used for subsequent sound analysis.

Analysis of data

The decibel readings in the row-column format provided a time-oriented record of the sound levels that existed at a given microphone. These readings, when considered individually, were valuable indicators of the sound environment in a particular area and were adequate to show maximum, minimum, and average levels. For this study, the use of computer developed sound contours provided a more comprehensive record of the variations in the laboratory sound environment that resulted from treatment changes.

Definition of Terms

Many of the definitions in this list have been taken from the glossary of the Public Health and Welfare Criteria for Noise (U.S. Environmental Protection Agency, 1973). The sources of other definitions will be cited or otherwise identified.

A-weighted sound level: The ear does not respond equally to frequencies, but is less efficient at low and high frequencies than it is at medium or speech range frequencies. Thus, to obtain a single number representing the sound level of a noise containing a wide range of frequencies in a manner representative of the ear's response, it is necessary to reduce, or weight, the effects of the low and high frequencies with respect to the medium frequencies. The resultant sound level is said to be A-weighted, and the units are dB. A popular method of indicating the units is dBA. The A-weighted sound

level is also called the noise level. Sound level meters have an A-weighting network for measuring A-weighted sound level.

Ambient noise: The all-encompassing noise associated with a given environment, being usually a composite of sounds from many sources near and far. (Olishifski & Harford, 1975, p. 1020)

Analog: If a first quantity or structural element plays a mutually similar role as a second quantity or structural element, the first quantity is called the analog of the second, and vice versa. (Olishifski & Harford, 1975, p. 1020)

AND gate: A circuit which has two or more input-signal ports and which delivers an output only if and when every input signal port is simultaneously energized. (Lapedes, 1974, p. 66)

ANSI: The American National Standards Institute, a standards-making body. (Olishifski & Harford, 1975, p. 1020)

Binary: Possessing a property for which there exists two choices or conditions, one choice excluding the other. (Lapedes, 1974, p. 159)

Decibel (dB): Objective unit of sound pressure level (p) defined with respect to the arbitrary zero for a sound pressure scale of $0.0002 \text{ dynes/cm}^2$, viz

$$\text{dB} = 20 \log_{10} p + 74 \text{ (ref } 0.0002 \text{ dyn/cm}^2\text{)}. \text{ (Warring, 1969, p. 73)}$$

Dosimeter (noise dosimeter): An instrument which registers the occurrence and cumulative duration of noise exceeding a predetermined level at a chosen point in the environment or on a person.

Loudness: An attribute of an auditory sensation in terms of which sounds

may be ordered on a scale extending from soft to loud. Loudness is chiefly a function of intensity but it also depends upon the frequency and waveform of the stimulus. The unit is the sone.

Loudness level: The loudness level of a sound, in phons, is numerically equal to the median sound pressure level, in decibels, relative to 0.0002 microbar, of a free progressive wave of frequency 1000 Hz presented to listeners facing the source, which in a number of trials is judged by the listeners to be equally loud.

Microbar: A microbar is a unit of pressure, equal to one dyne per square centimeter.

Microphone: An electroacoustic transducer that responds to sound waves and delivers essentially equivalent electric waves.

NAND circuit: A logic circuit whose output signal is a logical 1 if any of its inputs is a logical 0, and whose output signal is a logical 0 if all of its inputs are logical 1. (Lapedes, 1974, p. 985)

Noise: (1) Disturbing, harmful, or unwanted sound; (2) an erratic, intermittent or statistically random oscillation.

Noise exposure: The integrated effect over a given period of time of a number of different events of equal or different noise levels and durations. The integration may include weighting factors for the number of events during certain time periods.

NOR circuit: A circuit in which output voltage appears only when signal is absent from all of its input terminals. (Lapedes, 1974, p. 1013)

OR gate: A multiple-input gate circuit whose output is energized when any one or more of the inputs is in a prescribed state; performs

the function of the logical inclusive-or; used in digital computers.
(Lapedes, 1974, p. 1050)

Phon: The unit of measurement for loudness level.

Sound: (a) The auditory sensation produced through the organs of hearing usually by vibrations transmitted in a material medium, commonly air.
(b) Sound is an oscillation in pressure, stress, particle displacement, particle velocity, etc., in a medium with internal forces, for example, elastic, viscous, or the superposition of such propagated oscillations. (Olishifski & Harford, 1975, p. 1054)

Sound level meter (SLM): An instrument, comprising a microphone, an amplifier, an output meter, and frequency-weighting networks, that is used for the measurement of noise and sound levels in a specified manner.

Standing wave: Periodic wave having a fixed distribution in space which is the result of interference of progressive waves of the same frequency and kind; characterized by the existence of nodes or partial nodes and anti-nodes that are fixed in space.

Transducer: A device capable of being actuated by waves from one or more transmission systems or media and of supplying related waves to one or more other transmission systems or media. The waves in either input or output may be of the same or different types (electric, mechanical, or acoustic). (Olishifski & Harford, 1975, p. 1059)

CHAPTER II. REVIEW OF LITERATURE

Introduction

One of the forces of dynamic change exerted on mankind by modern technology is the continuous introduction of new and sophisticated machines of speed and convenience for use at work, in the home, and during leisure activities. While the positive attributes of these technical advancements are obvious, grave concerns are being raised about the wisdom of the technical choices made in our society. National Technical Information Service (1971), in the report HUD Noise Assessment Guidelines Technical Background, presented these observations:

One of the most conspicuous results of current technology is that it exposes us to noise: aircraft, automobiles, trucks, railroads, construction equipment, factories--even home appliances--all contribute to the din that characterizes modern cities. Like air and water pollution, most noise comes of our having made particular technological choices without fully considering their impact on the people who have to live with them. Technology, to date, has typically advanced by satisfying "first order" needs with "first order" solutions--for example, creating transportation facilities (the automobile and the highway system) to increase our mobility. It has responded to specific problems in isolation rather than anticipating the sociological effects of the solution, and has simply "built" systems, rather than designing them with an awareness of their potential overall impact on society.

This approach appeared at first to be adequate only because our natural resources of land, air, and water were so great as to be practically infinite, in relation to the existing demands. But with the growth of our population and the spreading and crowding of our cities, such conventional "first order" solutions have gradually come to defeat the purposes for which they were made: we now have traffic congestion instead of mobility, and also problems of air and noise pollution as well. (pp. 16-17)

In reviewing these observations, it is difficult to envision returning to primitive ways of life, therefore the inadequacies of previous "first order" decisions must be corrected and future decisions based on

"long range" consequences as well as those of the present. In the case of noise pollution, this has begun. The regulations for maximum noise exposure at the work place, as set forth by the Federal Occupational Health and Safety Act, have been adopted by many states, including Iowa, to apply to the public and private school laboratories. The enforcement of this law introduced the need for noise measurement in school laboratories and was the basis for this sound measurement research. In order to logically discuss sound measurement, an initial review of sound properties and the hearing process will precede the general discussion of sound measurement. This portion of the review of sound measurement literature will be followed by an introduction to the sound level meter and a discussion of the methodology used in school sound level studies.

Properties of Sound

Sound can be defined as a form of periodic energy that gives rise to the sense of hearing. While this definition is not inaccurate, it is of limited value because it does not convey any idea of how sound is generated and propagated.

Sound generation

The presence of sound depends on an oscillating or vibrating energy source or one that may be in the form of transient or periodic impacts. If the many possible types of sound generators were to be enumerated, the list would be formidable, but according to White (1975), most of them would fit into the following groups:

1. Vibrating solid bodies, for example, tuning forks, transformers, loud speaker diaphragms, stringed or percussion instruments, and

- sounds arising from complex components or structures.
2. Vibrating air columns, as exemplified by resonating organ pipes and wind instruments.
 3. Transient forms of mechanical or electrical power, such as volcanoes, avalanches, and lightning discharges.
 4. Impact phenomena, for example, tapping and hammering.
 5. Sounds from rapidly expanding gases, such as from jets, rockets, and chemical explosions.
 6. Complex sonic disturbances resulting from rapidly moving objects in fluids, such as fans, propellers, and shells. (p. 21)

The most common source of sound is the mechanical vibration of a solid surface. For example, if a vibrating machine is enclosed by sheet-metal panels, the vibration causes the panels to be set into motion back and forth against the surrounding medium. This process of coupling the sound energy of an object to its environment is fundamental to sound production. A familiar device involving coupling is the piano string. Vibrating by itself, a relatively small area reacts with the surrounding air, but with the addition of the sounding board that has a large surface to move the air, the vibrations of the string are amplified as a result of the improved coupling. The transmission of sound through the medium of air is naturally the most familiar, but it must be pointed out that sound will travel through almost any solid liquid or gas.

Propagation of sound

Sound propagation is a process that takes place within a material medium. It can be transmitted through solids, liquids, or gases, but not through a vacuum. Sound is a type of elastic wave motion. In order for a wave motion to exist, there must be a disturbance to the equilibrium of the material and, in addition, the disturbance must move

through the material with no net transport of the material of the medium. A disturbance can move through a medium in two ways: the vibration or maximum and minimum excursions of the disturbance (wave) can be lateral or transverse to the direction of movement of the wave or move in the same (longitudinal) direction.

Transverse wave motion The familiar waves that move outward from the point of impact when an object is dropped into a quiet pond are transverse because the water molecules are displaced perpendicularly to the direction of the wave propagation. This perpendicular displacement is caused by sheer or viscous forces within the water and are not accompanied by any dominant density or pressure changes in the medium.

Longitudinal or sound waves In contrast with transverse waves, the disturbance in longitudinal waves moves along the line of the direction of propagation. Sound waves are of this type with the disturbance being in the form of pressure changes, a series of alternating high and low pressure areas propagating through the medium. In order to better understand this phenomenon, consider the example of the vibrating machine enclosed in sheetmetal panels. As the vibration of the machine causes a panel to move back and forth, it will first move forward into the adjacent air molecules creating an area of pressure or compression. At normal atmospheric pressure, air is considered to be an elastic medium where coupling exists between adjacent molecules, therefore the compression produced by the forward movement of the panel results in a wave motion moving away from the panel. The actual wave is developed by the slight forward movement of the molecules next to the panel

transferred to the next along the line of propagation. During the second half cycle, the vibrating panel will move backward from the adjacent air causing a lowering of pressure or rarefaction which again will be passed along down the line of propagation. These small pressure variations generated by the group of molecules set in motion by the original sound source can be sensed and thus facilitate the detection of a sound wave.

A graphical representation of sound waves is shown in Figure 1.

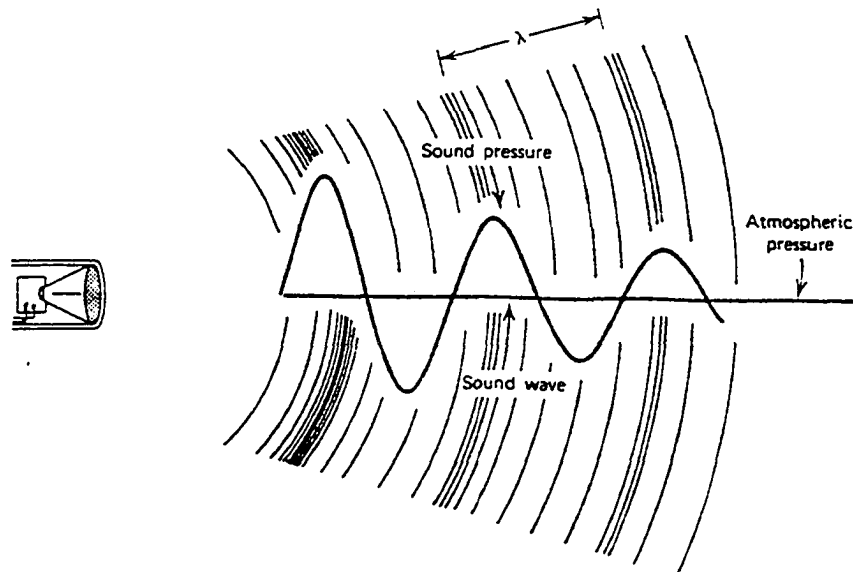


Figure 1. Pattern of compressions and rarefactions surrounding a point source. The distance between successive compressions and rarefactions characterizes the wavelength, λ (White, 1975, p. 27)

The closely spaced radiating concentric lines denote areas of increased pressure and the wider spacing indicates reduced pressure. The center transverse wave represents the air pressure changes around normal

atmospheric pressure that is due to the sound wave.

The moving sound waves in air can be one of two types--plane waves or spherical waves. The report on Fundamentals of Noise: Measurement, Rating Schemes, and Standards (1972) provided the following explanation:

Plane waves

For discussing sound waves in air, it is helpful to think about two types of waves--the plane wave and spherical wave. If sound propagates in one direction only, the forward edge of the wave lies on the surface of a plane perpendicular to the direction of propagation of the wave. Measurement of the pressure fluctuations associated with a plane, or free, progressive wave is simplified because the location of the transducer (a device which translates the changing magnitude of one kind of quantity, e.g., sound pressure on a microphone, into corresponding changes in another kind of quantity, e.g., voltage) is unimportant. The sound pressure is the same everywhere in space, except for relatively small effects due to absorptive and dissipative losses in the medium. An approximation to this simple kind of wave is obtained far away from a sound source when it is placed in an acoustically free field (a field without any nearby sound reflecting obstacles). (p. 1)

Spherical waves

Another type of sound wave frequently encountered in practice is the small spherical wave. Such waves can be thought of as waves propagating away from a small point source with an absence of reflecting surfaces in the vicinity of the source and the point of measurement. An instantaneous picture of the pressure distribution in the forward edge of the pressure wave would show that it lies on the surface of a sphere, with the center of the sphere acoustical intensity (sound power per unit area) of spherical waves decreases as the wave gets farther from the source, according to a relationship known as the inverse square law. In an ideal free field, and with no dissipation, a 6-decibel (dB) decrease in sound pressure level...could be expected for each doubling of distance. Since real conditions are not ideal, in practice some loss other than 6 dB per double-distance can be expected.

The inverse square law governs the intensity of the free sound radiation in the acoustical far field of a sound source. The acoustic far field is the region where the sound wave diverges as from a spherical source. If sound measurements are made in the far field, then the sound level farther from the source can be accurately predicted. (p. 2)

The square law states that the pressure of a spherical wave under far field conditions will vary inversely as the square of the distance from the source. By means of clarification, the far field is that distance from the sound source where the square law will reliably predict the sound pressure at a point a known distance from the point where the pressure has been established. At a certain distance from a point source of sound before the waves have stabilized to far field conditions, the square law will not be valid. This area around the source is known as the near field.

This concept of sound being a moving disturbance in an elastic medium serves as a foundation for the development of the descriptors that will make mathematical and other analyses possible.

Sound descriptors Physical happenings such as wave motions may vary greatly in form and function while being definable by common descriptors. For this study the amplitude, frequency, velocity, wave length, attack and decay characteristics, and spectrum were considered.

Amplitude Sound was described to be a pressure disturbance propagating through a medium. If the magnitude of the disturbance or particle displacement from rest position in the medium increases, the amplitude can be said to increase. The human ear is a sensitive transducer of pressure change to change in nerve impulses. The impulses are decoded by the brain to become what man perceives as sound. For a given circumstance within the physical capabilities of the human ear, an increase or decrease in amplitude will be perceived as being louder or

softer.

Sound amplitude can be described in terms of sound pressure level (SPL), sound power level (PWL), or sound intensity level (IL). All three methods use the decibel scale because of the great range to be covered. The decibel (dB) is a logarithmic scale whereby the ratio between an observed level is compared to a standard reference. The scale is meaningless unless the reference is known. The decibel concept can be explained by means of the transformation equation $X = \log_{10} Y$, where X is a logarithmic measure of Y . If X is known, the value for Y can be found from the reverse transfer equation $Y = 10^X$, where X is the exponent for expressing Y in the powers of 10. Originally, this logarithmic operation was called a Bel, but because the operation resulted in small numbers, one-tenth Bel or decibel was used, that is, decibels (dB) = $10 \log_{10}$.

Further clarification of the decibel scale can be made by examining the actual equations for expressing pressure, power, and intensity levels in terms of decibels.

1. $L_p = \text{SPL} = 10 \log_{10} \frac{p^2}{p_r^2} = 20 \log_{10} \frac{p_{\text{rms}}}{p_r}$, and $p_{\text{rms}} = p_r \frac{\text{SPL}}{10^{20}}$,
or $p^2 = p_r^2 \frac{\text{SPL}}{10^{10}}$
2. $L_I = \text{IL} = 10 \log_{10} \frac{I}{I_r}$, $I = I_r \frac{\text{IL}}{10^{10}}$
3. $L_W = \text{PWL} = 10 \log_{10} \frac{W}{W_r}$, $W = W_r \frac{\text{PWL}}{10^{10}}$ (Chanaud, 1972, p. 3).

The symbol $\overline{p^2}$ refers to mean square pressure. It is obtained by determining the average of the square of several pressure readings taken within a given time interval. Extracting the square root of the mean square pressure will give the root mean square (rms) or effective pressure. In this case p_r refers to a standard reference pressure of 20 micronewtons per square meter ($20\mu\text{N}/\text{m}^2$). This corresponds to the pressure at the threshold of normal hearing and is equal to 2.90065×10^{-9} pounds per square inch (psi).

Sound pressure level (SPL) is identical to L_p and equals the pressure (p) ratio as expressed in decibels. The term level implies that a reference level is to be considered.

Sound intensity (I) is defined as the sound power passing in a defined direction through a unit area. It is represented in watts per area and the standard reference level I_r is 10^{-12} watts per square meter ($10^{-12} \text{ W}/\text{m}^2$). Sound power is expressed in watts with the reference being 10^{-12} watts.

In regard to acoustic power, instrumentation for directly measuring the power level of a source is not available, but instead is computed from sound pressure measurements (Peterson & Gross, 1974, p. 7). The power of some common sounds is presented in Figure 2. It is important to note that actual acoustical power differs greatly from electrical power commonly used to describe or rate the power capabilities of sound reproduction systems. This discrepancy is apparent from Figure 2 that shows the sound power from a large orchestra to be 10 watts. Normally an amplification system capable of producing this volume of

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ACOUSTIC POWER

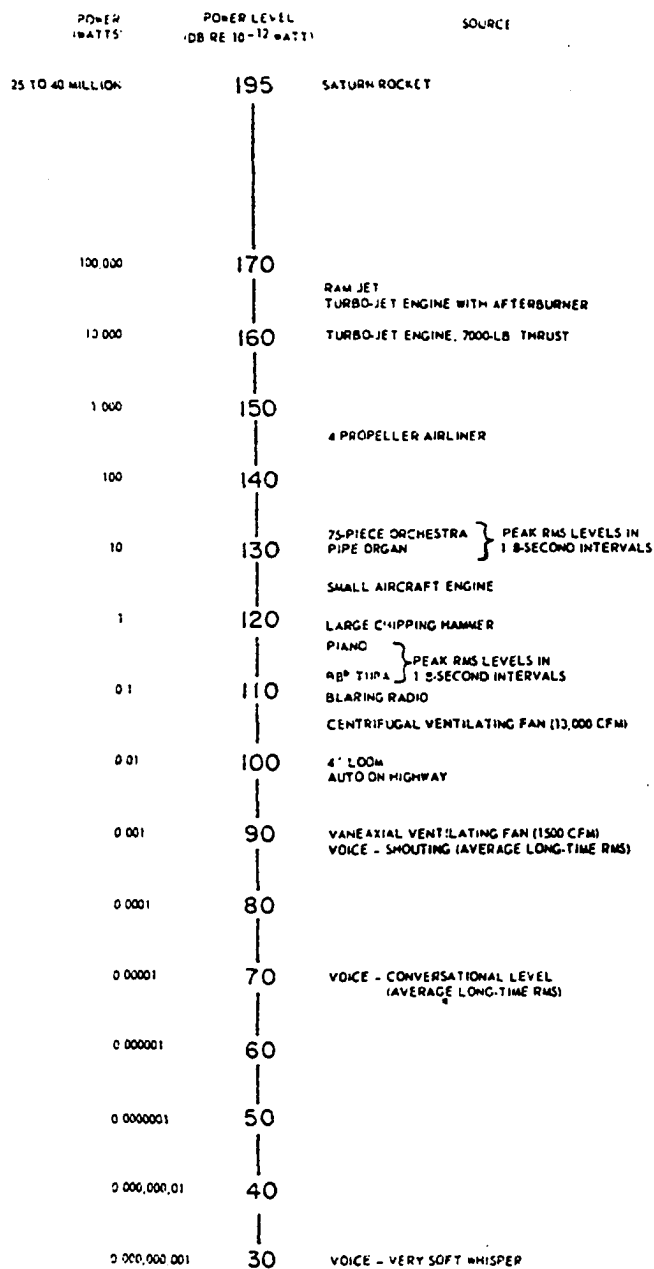


Figure 2. Typical power levels for various acoustic sources (Peterson & Gross, 1974, p. 6)

sound would require amplifiers rated much greater in electrical watts.

White (1975) stated:

The acoustic power of a loudspeaker may be only one watt even though the electrical power rating may be 40 watts, because of the inefficient process of converting electrical to acoustic energy. (p. 41)

One consideration of sound power that must not be overlooked is the addition or combining of sound sources. The combined power of two 60 dB sources is not 120 dB. Instead, the combined power level is 63 dB as can be determined from the following:

$$PWL = 10 \log_{10} \frac{W_1 + W_2}{W_r} = 10 \log_{10} \frac{2W_1}{W_r} = 3\text{dB} \quad (\text{Chanaud, 1972, p. 3}).$$

The power corresponding to the dB for each level is found and then added. The level in dB for the combined power is determined, and in the case of doubling, the increase is three dB. Avoidance of tedious calculations can be made by consulting a chart for adding dB as presented in Figure 3.

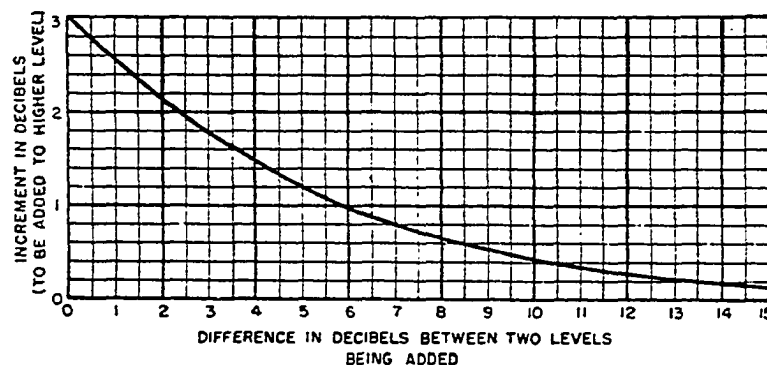


Figure 3. Chart for combining noise levels (Peterson & Gross, 1974, p. 9)

Frequency The frequency of a sound wave refers to the number of complete pressure change cycles that occur in one second. A tone

would be heard if a sound source were to cause a pressure increase to maximum which then returns through zero to normal pressure to a minimum value and then back to normal 1000 times per second. The frequency of this tone would be 1000 hertz (Hz) (Warring, 1969, p. 9).

Velocity A sound wave is propagated at a velocity dependent on the elastic properties of the medium. It is independent of frequency and in air at 0 degrees Celsius (C) is 331/45 centimeters per second (1087.42 feet per second). In accordance with the following formula

$$c = 33145 \frac{T}{273} \text{ cm/sec (White, 1975, p. 29)}$$

where c equals velocity in centimeters per second and T is the temperature in degrees Kelvin, the speed of sound is affected by temperature and has been found to increase with rising temperature.

Wavelength Wavelength is the separation along the direction of propagation between parts of the sound wave that are identical in phase of motion. In accordance with the formula

$$\text{wavelength (y)} = \frac{\text{velocity (c)}}{\text{frequency}}$$

wavelength is inversely proportional to frequency. The normal human ear, depending on age, has a frequency range of approximately 20 Hz to 16,000 (16 kHz), thus the wavelengths of major concern are from 17 meters to 2.1 centimeters (Warring, 1969, p. 9).

Attack and decay characteristics The character and effect of a sound is influenced by the manner in which it begins, continues, and ends with respect to time. In this regard, sound can be classified

as (1) ambient or background sound, (2) steady-state sound, (3) changing or intermittent sound, and (4) impulsive sound.

Ambient sound is environmental noise from distant, unassociated sources as well as local sources. It is usually at some minimum level that exists when specific sound sources under investigation are inoperative.

Steady-state sound exists where several machines operate with a high percentage of on time to create an overall continuous sound level.

An example of changing or intermittent sound is that of traffic or operations where a primary sound source is on and off at frequent intervals. The on period must be of greater duration than the normal time the ear requires to react to a sound.

Impulse sounds are characterized by brief periods of maximum pressure attained after a very short rise time, often in the microsecond range.

Sound spectrum If a sound wave of a certain frequency varies in a periodic manner identical to a sine wave, it is said to be a pure tone. Using this tone as a reference or fundamental, a similar tone of twice the frequency would be called the second harmonic. This relationship could be continued to create an infinite number of harmonics relative to a given fundamental tone. Working from this background, the French mathematician, J. F. J. Fourier, showed that any periodic wave form can be formed by adding a sequence of harmonics of the proper amplitude and phase relationship (White, 1975, p. 53). It must be emphasized that sounds of complex harmonic structure need not be

periodic or repetitive. Noise or unwanted sound as determined by the subjective processes of hearing may be made up of all kinds of sound to include periodic, aperiodic, impulsive, and broad bands of frequencies (White, 1975, p. 52). It is the general case that noise will be rather complex and will embody many frequencies and bands of frequencies.

The concept of bands of frequencies and bandwidth is of major importance when analyzing sound and noise. The term band of frequencies implies that a sound source or receiving device can respond to a spectrum of frequencies of varying intensities. Bandwidth is a general term relating to the frequency of a device.

The previously stated descriptors delineate the nature of sound; however, in analysis, certain interaction effects that occur when sound is propagated from one medium to another merit discussion.

Interaction phenomena Chanaud (1972) wrote about these effects:

The propagation of any disturbance through a change in the material media always results in three consequences: reflection, transmission and transduction. Transduction is a better descriptor than absorption because there are times, especially in buildings, where sound is transduced to vibration (locally absorbed) but later comes back to haunt you at some other location as sound and so it really was not absorbed in the proper definition of the word.

Reflection is a process in which the sound energy is forced to change direction. Plane reflecting surfaces can be used to redirect sound so that the wave fronts are of similar shape, concave reflecting surfaces cause focusing or intensification of sound pressure at certain places and convex reflecting surfaces result in spreading or scattering of incident sound. All of these effects are used in practical acoustics to achieve an end, sound amplification in an auditorium or noise reduction. Transduction occurs in two forms. One is when the energy is converted to heat (resistive absorption) and becomes unavailable for sound. The other is when vibration is induced in another medium that later may become available for sound production (reactive absorption is one descriptor

used for part of this process). Both transduction processes generally are strongly frequency dependent and so alter the source spectrum. In noise control we use these characteristics to reduce the received sound in the audio spectrum of frequencies.

Another propagation effect which can be important is refraction. Consider a sound wave propagating horizontally over the ground whose air is warmer close to the ground. Since the speed of sound in air increases as the square root of the absolute temperature, the sound closer to the ground will propagate faster. Before the temperature difference is encountered, the locus of maximum sound pressure (a wavefront) will be a vertical line but will move horizontally at the same speed. Upon encountering the temperature gradient, the sound speed will increase near the ground and so after a given time the wavefront near the ground will be ahead of the wavefront that was once directly above it. The wavefront will be curved upward. Since the energy transfers perpendicular to the wavefront, it must now have a vertical component, i.e., the sound energy begins to rise, creating a lesser amplitude sound field on the surface. Such "shadow" zones explain why it is difficult to speak across hot fields in the summer and conversely why sound carries remarkably across a lake in the evening where the surface is cooler than the air and the sound bends downward and stays on the surface.

Diffraction is another effect which can influence the sound level and frequency distribution at distant points. Partial height barriers used in open plan offices or dirt berms used on highways are places where sound diffraction is important and so are useful in explaining this phenomenon. Waves have the ability to bend around objects the longer the wavelength with respect to the size of the object, the greater this bending can be. Thus a partial height barrier can provide shielding for frequencies whose wavelengths are less than the height of the barrier but are decreasingly effective for progressively lower frequencies. The same phenomenon occurs with ocean waves; one never sees height differences around jetties due to the tidal variation (very long wavelengths) but these differences are obvious for wind generated waves (shorter wavelengths). (p. 5)

The previous text has presented basic background information concerning sound and its propagation in an environment of free space.

Reference was made to a free field as being an area surrounding a source of sound that resembles open space. Under these conditions a sound from a point source would be in the form of spherical waves and the intensity

from the source would vary in accordance with the inverse square law. Within an interior environment or enclosed area such as that of an industrial arts laboratory, free field conditions cease to exist due to the various interactions of the sound with the equipment and interior surfaces. This multisource, direct, reflected, diffracted, and superimposed sound is thought of in terms of sound field rather than an individual sound wave. The sound field within an enclosure, depending on its physical characteristics, may be termed as diffuse and reverberant.

Diffuse field A diffuse field will be created if a large number of reflected or diffracted waves combine to cause the sound energy to be uniform within the enclosure. In an enclosure with reflective surfaces a reverberant field will exist.

Reverberant field Reflected sound that continues in a time-dependent manner after the cessation of the sound source causes the reverberant field. An enclosure with surfaces of high reflective efficiency will produce a reverberant field such that the ambient noise level resulting from multiple noise sources will be high.

The Ear and Hearing Process

The introductory discussion has been concerned with the subjective aspects of sound and sound measurement, the major emphasis of this study. Central to any undertaking relative to sound is the receptor through which man is able to perceive sound, the human ear. A study regarding sound measurement must include the ear and the hearing process.

The simplified drawing of Figure 4 shows the three regions of the ear: outer, middle, and inner. The outer ear is made up of the visible

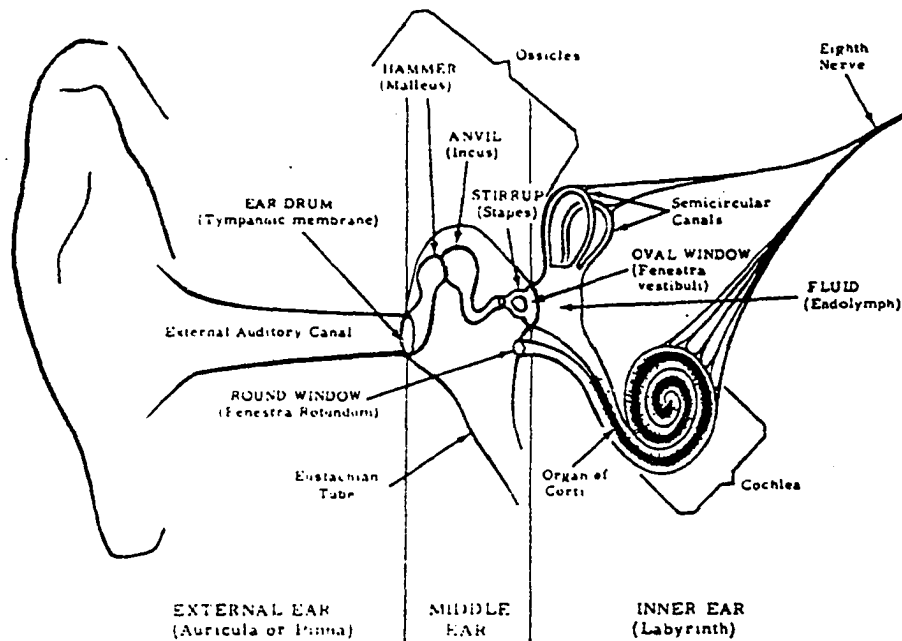


Figure 4. Shown here is a schematic diagram showing the transmission of sound waves to the inner ear (Olshifski, 1975, p. 12)

part or pinna and the auditory canal which is terminated by the tympanic membrane or ear drum. The middle ear is the space on the inner side of the ear drum and is occupied by three small bones called the auditory ossicles. Pressure differentials between the outer and middle ear due to changes in atmospheric pressure are relieved by the eustachian tube which leads from the middle ear to the throat. The inner ear consists of fluid-filled passages within the skull bone and is called the cochlea. One set of these passages known as the semicircular canals is part of the balance detecting mechanism and is not directly connected with the hearing process. The cochlea is shaped like a snail shell coiled about two and one-half times its own axis. If unwound it would be similar to an elongated tube partitioned into upper and lower galleries filled with

fluid and separated by the basilar membrane with its associated cochlear duct and organ of Corti as shown in Figure 5.

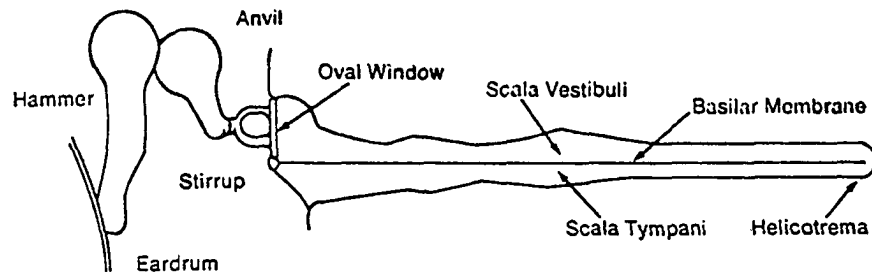


Figure 5. Here is how the cochlea would look if it were uncoiled (Clemis, 1975, p. 218)

The Hearing Process

A sound wave as gathered by the pinna is converged through the auditory canal to the ear drum. In a manner similar to the diaphragm in a microphone, the ear drum vibrates in response to the pressure changes due to the presence of the sound. As shown in Figure 4, these vibrations of the ear drum are mechanically conducted by the ossicular chain or bones of the middle ear to oval window membrane of the cochlea. As the result of the lever action of this ossicular chain and the larger relative size of the ear drum, the pressure sensed by the ear drum is magnified at the oval window. The pressure variations at the oval window pass through the fluid of the gallery identified as the scala vestibuli to the apical ending of the coil known as the helicotrema and on to the other gallery. This gallery, the scala tympani, is terminated by the membrane of the round window which responds in a manner opposite to the movement of the oval window. This resilient action of the membrane

in the round window is necessary to relieve the fluid pressure, otherwise the oval window membrane could not move in response to the vibrations transmitted by the ossicular chain. In conjunction with associated membranes, the organ of Corti with its hair cell terminations of the auditory nerve physically separates the scale vestibuli and the scala media. The pressure waves induced in the fluid of these passages cause a sheer pressure to be exerted on the hair cells of the organ of Corti. This stimulation sets up the electrical impulses transmitted by the auditory nerve to the brain. The method by which these impulses are decoded to produce the sensation of sound is still a matter for research; however, experiments to develop scales of loudness such as the phon scale in Figure 6 convey a general idea of the overall response of the ear.

While the ear is a sound recognizing device of considerable sophistication, subjective experiments are the most practical way to determine its characteristics. An approach in this direction is the development of the phon and sone scales.

Phon scale The family of curves presented in Figure 6 were developed experimentally and are the result of many observations whereby listeners were asked to judge when a pure tone of a given frequency sounded at the same loudness level as a 1000 Hz reference tone. The reference tones are selected at 10 dB (re 20 uN/m^2) intervals starting with zero dB at the minimum audible field (MAF) and extending to 120 dB. These 1000 Hz reference levels represent units of loudness level in phons. It can be noted from Figure 6 that a 100 Hz tone at a sound pressure level of 60 dB has a loudness level of about 50 phons, or a

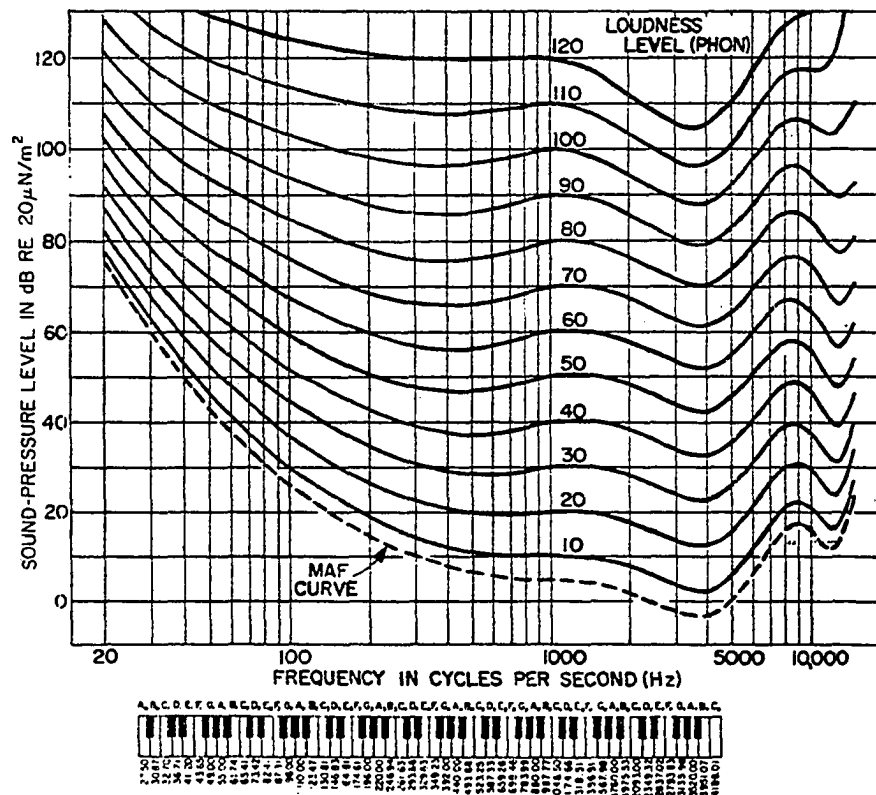


Figure 6. Free-field equal-loudness contours for pure tones (observer facing source), determined by Robinson and Dadson 1956 at the National Physical Laboratory, Teddington, England. (ISO/R226-1961) Piano keyboard helps identify the frequency scale. Only the fundamental frequency of each piano key is indicated (Peterson & Gross, 1974, p. 21)

1000 Hz tone at a sound pressure level of 50 dB sounds as loud as a 100 Hz tone at a sound pressure level of 60 dB. It should be noted at this time that human perception of sound is a subjective process and many scales and measurement methods have been developed and each approaches the subject in a specialized way.

Sone scale Starting with a given sound level, it has been found that a 10 phon increase in loudness level appears to the average ear as having doubled in loudness. The sone scale is a linear loudness scale

based on this relationship such that a doubling in sones represents a doubling in loudness. Since this is a scale of relative loudness, the reference level must be known. In general practice, a 1000 Hz tone with a sound pressure level of 40 dB re 20 $\mu\text{N/m}^2$ (a loudness level of 40 phons) is the standard reference and is equal to one sone.

The minimum audible field curve in Figure 6 represents the threshold of hearing for a normal young person. The ear, like other organs of the body, can be affected by certain factors such that a loss in sensitivity or reduction in ability to hear is the end result.

Hearing loss

Normal hearing ability can be affected in many ways. A loss in hearing is referred to as a threshold shift (TS) of a certain number of dB from normal. If an individual's hearing is reduced for a period of time and then returns to normal, this change is called a temporary threshold shift (TTS). In so many situations the ear is unable to rebuild or repair itself and a permanent loss in hearing may be incurred. This is known as permanent threshold shift (PTS).

Sound Measurement

An instrumentation system for measuring airborne sound will include: (1) a transducer, (2) electronic circuitry for either recording or processing the output of the transducer, and (3) provision for reading out the results.

Transducers

Sound measuring devices depend on a transducer to change the variations in pressure into corresponding electrical signals which will eventually result in an intelligible readout. When measuring airborne sound waves, the transducer is a microphone. The operation of this device centers around a sensitive diaphragm which is mechanically displaced by the sound pressure variations entering the microphone. In some manner, depending on the type of microphone, the movements of the diaphragm are converted to electrical impulses. Designers of present day sound measurement systems may use ceramic, condenser, electret or dynamic microphones.

The diaphragm of the ceramic microphone is mechanically connected to a piezoelectric crystal element that produces an electrical signal when a force is exerted on it. This is a reliable type of microphone that has been in use for many years. The main advantages of this microphone include ruggedness, immunity to the effects of humidity, and high electrical capacitance. A rather rough and limited frequency response plus the temperature limitation of the ceramic element and the adhesives used in its construction are the major disadvantages of this instrument.

As its name implies, the diaphragm of the condenser microphone is one plate of a capacitor (condenser is a lay term for a capacitor) which is subjected to a polarizing voltage. The diaphragm is mounted in close proximity to a stationary plate and the electrical output results from the change in capacitance caused by the movement of the diaphragm. This microphone has long-term stability characteristics, a broad frequency

range, and smooth response. Its shortcomings include the need for an external polarizing voltage, susceptibility to humidity, low electrical capacitance, and damage from shock such as dropping on a hard surface.

The electret microphones are a comparatively recent variation using the condenser principle. The diaphragm is made of plastic with a metalized coating on the external side. During the manufacturing process, the plastic diaphragm is permanently polarized, thus eliminating the need for the external polarizing voltage. Also, the nonconducting properties of the internal side of the plastic diaphragm make it possible for it to actually contact the stationary plate without reducing performance. As a result, the close tolerances inherent with the regular condenser microphone are reduced. Kamperman (1972) stated that "the electret condenser microphone appears superior to the ceramic microphone in all respects. However, the electret condenser microphone does not have as smooth a frequency response at high frequencies as the conventional condenser microphone, and it has a maximum temperature limitation of 50°C" (p. 87).

Dynamic microphones are designed with the diaphragm fastened to a coil located in a magnetic field, and movement of the diaphragm induces a voltage in the coil. The overall frequency response of this microphone, especially at low frequencies, is inferior to the other types. In addition, it is sensitive to extraneous alternating fields produced by power lines, electric motors, and transformers. Because of these disadvantages, the dynamic microphone is seldom used for sound measurements (Olishifski, 1975).

Electronic circuitry

The output of a microphone is in the form of minute variations in electrical potential or voltage. These signals are electronically processed for recording or to provide meaningful data readout.

Recording Many sound measurement and analysis procedures, especially those of an extensive nature, cannot be performed in the field because of logistical or environmental problems. A practical alternative is to tape record the sound at the site for analysis in the laboratory. This procedure was used by Large and Ludow (1975) to assess the reaction of an urban community to construction noise. The survey involved 193 tape recordings made from 43 locations within the selected area in North London, England. It should be noted that this survey used amplitude modulation recording directly on the magnetic tape. Because of the broad variations in level and frequency encountered in noise measurement, this type of recording requires a high quality recorder. Kamperman (1972) commented in this regard:

High quality audio recorders for broadcast use and High-Fi systems for the home can be adapted for recording acoustical data. However, this should only be done by personnel very experienced with the inherent limitations of tape recorders. Acoustical data generally covers a very wide dynamic range. Often there are transients and peaks that one would like to store on magnetic tape. Most professional broadcast recorders are not suitable for this application. They usually are lacking in adequate low frequency response, adequate dynamic range at low frequencies, excessive pre-emphasis at high frequencies (causing high frequency overload of acoustic signals) and most importantly they lack the indicator to show that transient or peak events have been faithfully recorded. (p. 94)

An alternative in recorder design which is used extensively is the frequency modulation (FM) system. This method utilizes a fixed amplitude carrier which is varied in frequency according to the sound signals

developed by the microphone. The advantage of this system rests with the fact that the constant amplitude carrier is selected to be within the magnetization range of the recording tape, thus eliminating the problems of amplitude distortion. In other words, the analog sound level information is in the form of a change in frequency, and by proper design this system can accommodate a greater range of input levels without distortion.

Signal processing for readout The small electrical signal from the transducer must be amplified to a level sufficient for analysis and readout. As a general rule, an instrumentation system will have a calibrated attenuator to divide the broad measurement range into adjustable steps. This increases the accuracy of the system because the readout is required to cover a small portion of the total range for a particular range setting. Also, this section of the instrumentation will include circuits to analyze and measure the characteristics of a particular sound.

Sound analysis Sound has been defined as a periodic wave of pressure change moving in an elastic medium. From this definition it can be implied that sound is a dynamic phenomenon with certain measurable qualities. Peterson and Gross (1974) discussed sound analysis as follows:

Electronic techniques can provide more information about sound or vibration signals than merely the over-all levels. We can find out how the energy of a signal is distributed over the range of frequencies of interest, a process that we can describe as analysis in the frequency domain. We can find relations among signals as a function of time by correlation techniques, and we can enhance the appearance of coherent elements in a signal, if a synchronizing trigger is available, by waveform averaging. These two we can class as analysis in the time domain. We can find the amplitude distribution of a signal, which shows how often the signal is at any of a possible

range of values, and this process we class as analysis in the amplitude domain. (p. 73)

Readout The final outcome of any measurement system rests with the readout section where the existence of a certain phenomenon is quantified in terms of some accepted scale or standard. The choice of readout device depends on the type of sound or noise to be analyzed and the method of analysis. The selection will include analog and digital meters, graphic level recorders, the statistical distribution analyzer, oscilloscopes, and computers.

Meters are one of the most common indicating devices and are used extensively in portable and hand-held equipment. According to Kerbec (1972), meters are typically useful for indications which tend to be steady in nature. These indications may be average, root-mean-square (rms), or peak values. Also, peak-hold meters are available which will hold or constantly display the maximum excursion of an impulsive or intermittent signal. It should be pointed out that meters, and especially analog types with inherent mechanical inertia, do not indicate a real time or instantaneous happening. This type of display can be best displayed by a cathode ray oscilloscope.

The graphic level recorder is a type of recording meter with the outcome being recorded on a moving chart in terms of decibels or some other appropriate scale. While the slow response of this device makes it inappropriate for recording impulse or rapidly changing effects, it will show amplitude variations over extended time periods. Also it is effective when used synchronously with a set of band pass filters to provide a strip chart record of sound levels in various frequency bands,

As described by Broch (1971, pp. 72-73), the statistical distribution analyzer is used in conjunction with the graphic level recorder. Electrical contacts, one for each of twelve counters that make up the readout portion of the instrument, are scanned by the recorder's writing arm. In this manner, the range of the device is divided into 12 intervals, and the time the writing arm is located in each interval is recorded on the counter for that interval. The final outcome of the analyzer is a 12-bar histogram of the relative sound levels included in a given sampling period.

The oscilloscope makes use of the low inertia and instant response characteristics of the cathode ray tube electron beam and, as a result, is an outstanding real time indicator. While the moving electron beam is dynamic in nature, storage oscilloscopes are available, making it possible to hold a waveform for photographic or other reproductive purposes. With this technique, not only the amplitude of the sound signal is identified, but also the duration and attack-decay characteristics can be observed.

The tabulation of data generated by many sound monitoring situations is for all practical purposes impossible without the computer. Kerbec (1972) commented:

When a large amount of data must be analyzed, a digital computer becomes almost a necessity. For example, assume that automobile passby noise of approximately a 5-second duration is available for analysis. If the frequency range of interest is 100 to 10,000 Hz (21 one-third octave bands between 100 and 10,000 Hz) and a frequency spectrum is desired every 0.5 second, there would be approximately 210 pieces of information (21 one-third octave bands times 10 interrogations during the duration of the passby) for every passby. If many automobiles were tested at various speeds, loads, pavement surfaces, etc., the amount of data points would number in the millions.

Such a measurement and analysis program would require a computer to efficiently handle the data. (p. 17)

Sound Level Meter

The basic instrument used for sound level measurement is the sound level meter, a compact portable device which incorporates the transducer, amplification-weighting-attenuator, and readout sections as shown in Figure 7. In fundamental physical terms, the sound level meter measures

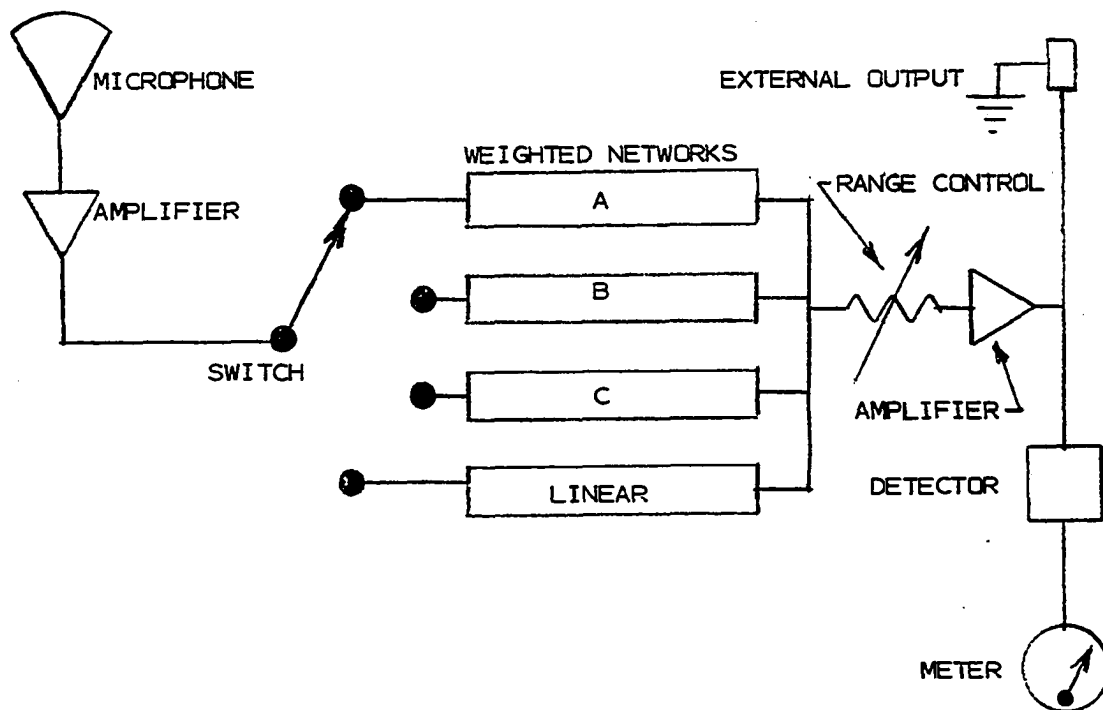


Figure 7. Block diagram of basic elements of a sound level meter

the sound pressure at its microphone. Such a measurement, however, would be more or less meaningless because it would not give an indication of the frequencies that make up the sound. Also, there would be no information as to the human perception of the sound. This concept of human

perception is of fundamental importance in sound and noise measurement and deserves further clarification.

Loudness level

The human ear is not a linear device. While the average normal hearing range is from 20 to 16,000 Hz, response or sensitivity is much greater in the range from 300 to 6,000 Hz. This discrimination of the human ear causes the sounds of different frequencies at identical sound pressure levels to be perceived by the average listener at varying loudness levels. Many psychoacoustical experiments have been conducted in an effort to develop a method to rate the loudness of a sound, and the phon scale shown in Figure 6 was one of many scales and measurement methods developed in this manner. In order to bring about a universal measurement approach, the internationally standardized sound level meter has been adopted. This meter is supplied with frequency weighting networks, the characteristics of which have been termed A, B, and C. The use of these networks is a relatively simple means of altering the frequency response of the sound level meter so that the measured results are more meaningful.

Weighting networks

Weighting networks are electronic circuits incorporated into the sound level meter to shape the frequency response of the device. The curves in Figure 8 show the A, B, and C frequency characteristics. All of these weighting methods may be used in the field of sound and noise measurement, but OSHA regulations, as set forth in Table 2, are indicated

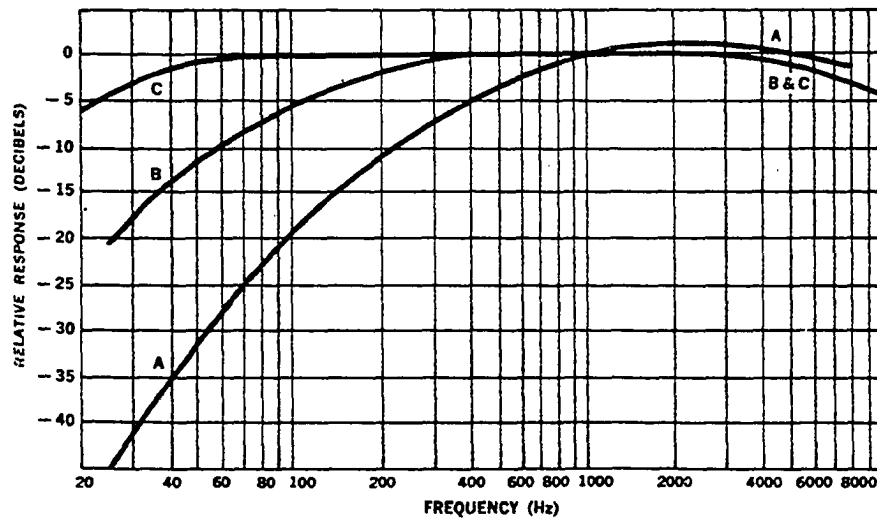


Figure 8. Frequency-response characteristics of a sound level meter with A-, B-, and C-weighting (La Jeunesse, 1975, p. 111)

in reference to A-weighted measurement. This research study was concerned only with A-weighted measurement.

The reasoning behind the A-weighted measurement becomes more apparent when comparing the A-weighted response curve of Figure 8 with the response of the human ear set forth in Figure 6. It can be seen that a rough similarity exists. A consideration that should be stressed at this time is the relationship between sound pressure meters and sound level meters. Similarly, each instrument measures sound pressure at its microphone, but by international agreement, when weighting networks are used, the outcome is termed as sound level. Referring again to Figure 7, if the linear setting is used, as is often the case when other analyses are desired, the instrument is then considered to be a sound pressure meter.

When making sound level observations with weighting networks, it is necessary to identify the type of network when recording the results. By

convention, a weighting network is identified by adding the letter for the network used either in text or to the decibel symbol. For example, in the case of an A network, the text would simply state: the A-weighted reading is __dB, or in symbol form, __dBA.

Sound level meter readout

The output of the amplification weighting and attenuator section is an alternating current (AC) signal which varies according to the changing sound pressures sensed at the microphone of the sound level meter. This signal passes to the readout section of the meter and is often made available in both weighted and unweighted form at accessory sockets for external processing. Within the readout section, the AC signal is detected or converted to direct current (DC). This detected signal drives a meter with scales calibrated in dB for each meter range. One important problem to be considered when using a meter for readout is the response time of the meter.

Many sounds are impulsive in nature with intermittent high pressure levels of very short duration. An instantaneous reading made during the brief time when the level was high, would give a false indication of the effective sound energy present. The readout meter can be made to integrate or average this type of sound energy by varying the response time of the readout meter. In accordance with American National Standards Institute specifications (ANSI S1.4-1971), sound level meters have two response positions, identified as slow and fast.

Response time According to La Jeunesse (1975), the slow response characteristics as specified by ANSI S1.4-1971 are such that a 1000 Hz

signal of 0.5 second duration applied to a Type 2 instrument will cause a reading of two to six decibels less than that of a steady signal. When the fast response setting is used, the reading will be zero to two decibels less than the steady signal. Also, this specification states that the decay time be essentially the same as the rise time and the reading for a steady signal in each position must vary no more than 0.1 decibel.

Types of sound level meters

Several types of sound level meters are available, and in the interest of maintaining uniformity between instruments produced by different manufacturers, the ANSI specification S1.4-1971 stipulates the performance guidelines for each type. The types of sound level meters are identified by Peterson and Gross (1974):

Four types of sound-level meters are specified in the latest American National Standard Specification for Sound-Level Meters, S1.4-1971. These are called Types 1, 2, 3, and S or "Precision," "General Purpose," "Survey," and "Special Purpose," respectively. The first three types differ in their performance requirements, with the requirements being most strict for the Type 1 or Precision Sound Level Meter and progressively less strict for the types 2 and 3. The special-purpose sound-level meter is one that meets the requirements of one of the other types, but does not contain all three weighting networks. (pp. 116-117)

An example of a common special purpose sound level meter is the personnel noise monitor or noise dosimeter. This is a small pocket-size device that is worn by the worker. It has a microphone and electronic circuitry to indicate the percent of total allowable noise exposure for a given time period in accordance with OSHA regulations. Also, this instrument is equipped with a warning light to indicate an exposure level

of 115 decibels and above during the monitoring period.

Monitoring Sound Levels in the Public Schools

A review of the literature reveals that extensive noise level studies with sophisticated instrumentation have been made in manufacturing, transportation, and urban settings. In contrast, the relatively few studies made in school laboratories relied on measurements made with hand-held sound level meters and dosimeters worn by instructors.

The procedure used by Hicks (1974) in a study of 30 Utah school laboratories employed a noise dosimeter worn on the person of the instructor during the normal school day and a Type 2 hand-held sound level meter to measure the A-weighted sound levels around the selected machines. In addition, the sound level meter was used to measure general environmental noise in open areas where audio communication was necessary.

A study of selected vocational laboratories in southeastern Michigan, Monfette (1974), utilized a Type 2 sound level meter to perform a preliminary measurement of general noise levels in a laboratory, a Type 1 sound level meter for more specific measurements as dictated by the preliminary survey, and two noise dosimeters to indicate cumulative noise exposure. All readings were made using the A-weighted setting designated as dBA.

The preliminary survey was made with the Type 2 meter being held waist high at arm length while the surveyor walked slowly about the laboratory once on the left side, once on the right side, and down the middle. The arithmetical mean for these three readings rounded to the next higher decibel was recorded.

If sound levels of 80 dBA were measured during the preliminary

survey, a second phase procedure involving the more precise Type 1 sound level meter was used. This instrument was set to respond to a frequency of 1000 Hz and the dBA readings at this frequency were recorded for as many activities at each work station as possible. These readings were averaged and rounded to the next highest decibel and recorded. This procedure was repeated for a minimum of five and a maximum of eight work stations within a given laboratory. The selection of the 1000 Hz frequency was based on the determination by the researcher that this frequency causes the greatest disruption in verbal communication.

The need for a third phase measurement procedure using the two dosimeters was dependent on the readings noted in phase two. If it was determined that a student work station activity was at 95 dBA or above, the instructor would wear a dosimeter. This five dB difference was based on the idea that students are in the laboratory for a shorter time than instructors, and higher exposure for students would cause OSHA violations in a shorter period of time.

A noise survey of selected industrial arts wood laboratories in the New Jersey public school system, Pinder (1974), was conducted to determine the noise levels produced by the machines regularly used in these laboratories. The instrumentation consisted of a Type 1 sound level meter mounted on a stand four feet from the floor and three feet from the machine or source of noise. The mounting of the meter on a stand eliminates reflections from the operator and reduces measurement error.

Jacobsen and Shadowens (1977) used a sound level meter (SLM) and a dosimeter in surveying the noise level in a university wood technology

laboratory. The two-hour laboratory sessions were divided into 10-minute sessions with measurement taking place only during laboratory practicums. For the first part of the experiment the 10-minute measurement segments were selected by use of a random number table. Sound level measurements were made with a hand-held SLM with the microphone at operator head height. For the second part of the experiment, two students were equipped with noise dosimeters during the two-hour class sessions. The dosimeters were read after each class session and the percent of total daily noise with respect to OSHA standards were recorded.

As a means of developing an informational background from which to approach sound measurement, this review of literature has identified the relationship that exists between sound, the hearing process, and sound measurement. Sound has been physically described as a varying pressure wave moving through an elastic medium; however, from the sensual standpoint, it is the human ear that reacts to these pressure variations to manifest through the sense of hearing, the phenomenon known as sound. Studies of the human ear describe it as being a sensitive pressure measuring device with decoding processes which are not totally understood. It is interesting to note the analogous approach in the design of sound measurement systems as compared to the human ear. An example is the sound level meter which uses a microphone and electronic circuitry to perform in a similar manner the function of the ear drum, middle ear mechanism, and partially that of the receptor in the inner ear. Finally, the results of the hand-held sound level measurements found in the school laboratory studies confirmed that machines known to produce excessive

noise when used in industrial settings were also noisy in school laboratories and that sound levels in excess of OSHA standards were encountered. Since the concern for the effect of noise pollution on the environmental quality of school activities has prompted the enactment of laws to include schools for compliance with OSHA regulations, it was deemed necessary for school administrators to base their decisions for corrective measures on the more extensive data collected by automated means. It was on this premise that this sampling and recording system was developed.

CHAPTER III. PROCEDURE

The objective of this study was to develop a sound measurement system capable of making measurements at several locations within a laboratory at precise intervals over an extended period of time. It must be emphasized that this research was not concerned with the development of a sound level meter. Instead, a commercially available and ANSI-approved instrument was used as the measurement standard for this system, with additional instrumentation developed to provide the means for automated multi-locational sampling and data recording for computer processing.

The procedure for this study was divided into four parts: (1) design and fabrication of the instrumentation, (2) testing and initial calibration of the instrumentation, (3) collection of data, and (4) analysis of data.

Design and Fabrication of the Instrumentation

The design and fabrication phase of the study was concerned with four general areas: (1) selection of a suitable sound level meter, (2) selection of 16 microphones, associated cable and connectors, (3) design and fabrication of a solid state switcher-converter, and (4) design and fabrication of an eight-level punch tape unit.

Selection of the sound level meter

The sound level meter was selected on the basis of three requirements: (1) it would be an ANSI-approved Type 2 or better instrument; (2) the design would provide for removal of the microphone for remote operation; (3) it would be compatible with the instrumentation in all

ranges from 80 dB to 140 dB.

These requirements were met by the Simpson Model 886 sound level meter. This instrument is an approved Type 2 meter with provision for microphone removal through the use of readily available Switchcraft Type A3 F and A3 M connectors. The meter reads from 40 to 140 dB in nine ranges and, most important for this study, an OSHA range from 85 to 115 dB is provided.

Selection of the microphones and cables

The Simpson sound level meter uses a detachable Type L condenser microphone which complies with ANSI specifications S1.12-1967. These microphones were available on separate order and 16 were obtained for use in the microphone matrix. The cables were made from Belden Number 8761 audio line and terminated with Switchcraft A3 F and A3 M connectors. These cables were electrically similar to the standard 25-foot cable furnished as part of the sound level meter package, and lengths up to 50 feet proved to be satisfactory for the system. The use of 50-foot cables without a pre-amplifier was of concern; however, experimental A-weighted readings of machine noise with and without a 50-foot cable revealed no appreciable difference in readings except for normal attenuation of the cable which was nullified by calibration.

Design and fabrication of the switcher-converter

The switcher-converter unit served three purposes: (1) it switched the microphones in sequence at timed intervals; (2) it converted the analog voltage from each microphone at a specific reading time to digital

form for the punched tape output; and (3) it made provision for counting events and designating microphone positions for calibration and normal operation.

Microphone switcher It can be noted from the block diagram of the system (Figure 9) that the microphone lines terminate at the switching relay block. These lines enter the switcher through the surface-mounted A3 M connectors as shown in the back panel layout (Figure 10). The relay block consists of 16 (one for each microphone) three-pole, single-throw reed relays, the three-pole configuration being necessary to accommodate DC voltage, audio, and calibrator switching for each microphone.

Switching relays The handcrafted relays consist of a copper bobbin with three center holes wound with 3000 turns of Number 30 copper magnet wire. Inductive transients produced by the coil are diode-dampened and switching is accomplished by glass-encapsulated miniature reed relays located in the center holes of the bobbin. The metal center of each bobbin acts as an audio shield and is connected to the shield of its respective microphone line. All centers terminate at one common point to eliminate ground loops which would introduce serious noise in the signal system.

The audio signal from each microphone enters the switcher through pin one of its respective A3 M connector located on the switcher back panel and terminates at an input reed switch terminal. The output terminals of the audio switches are commoned and lead through a shielded line to pin one of the lower A4 M connector on the back panel. A utility cable with an A4 F connector on one end and an A3 F on the other plugs

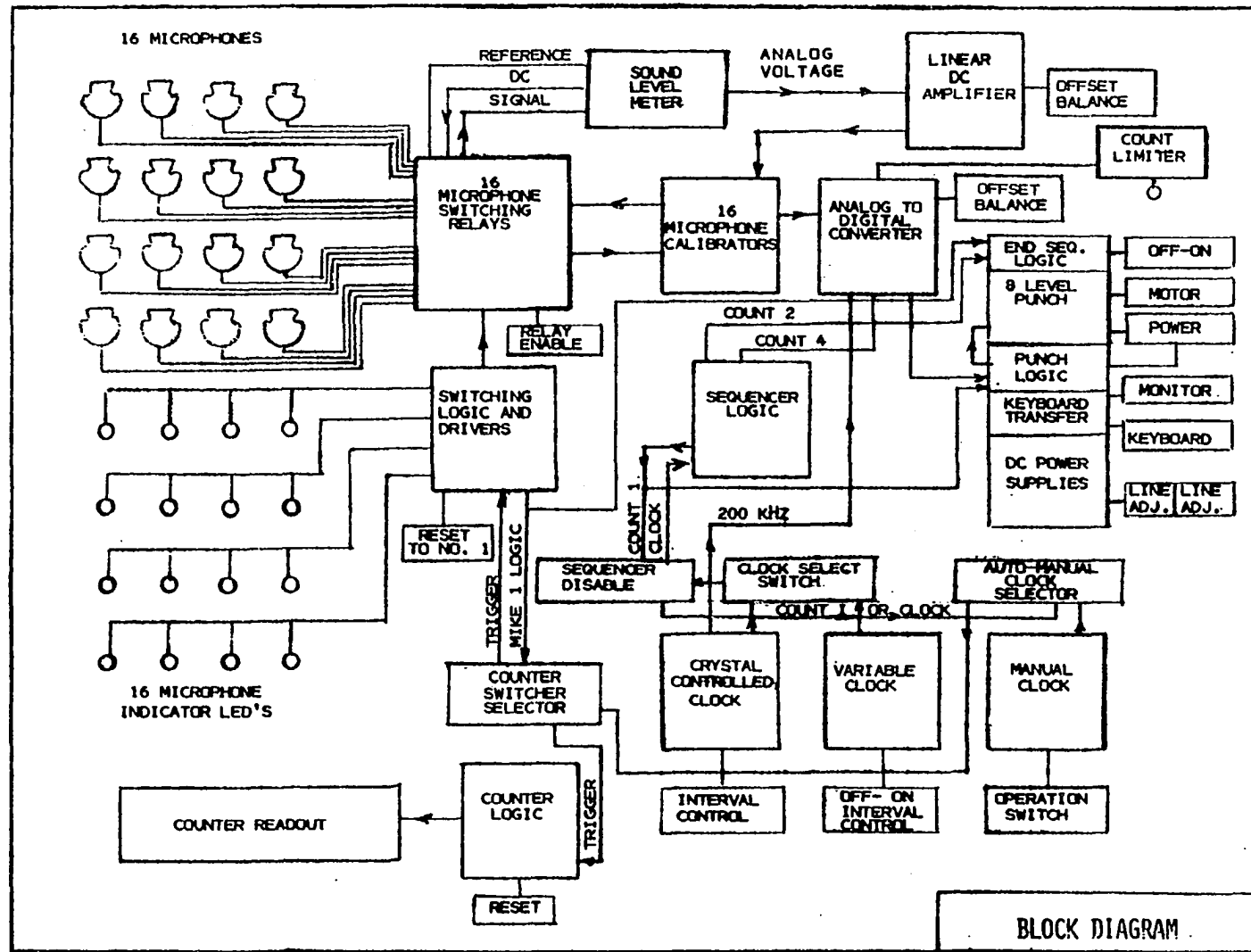


Figure 9. Block diagram

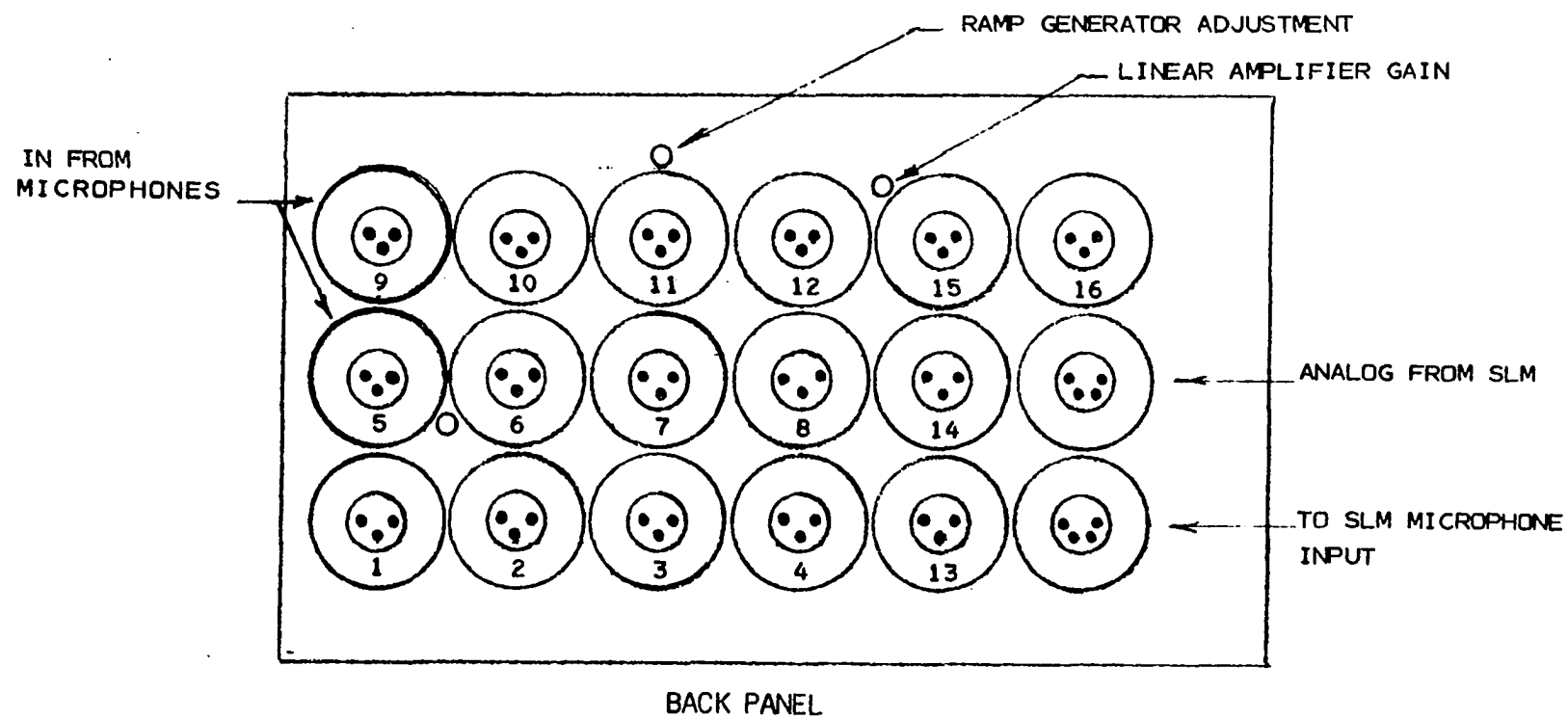


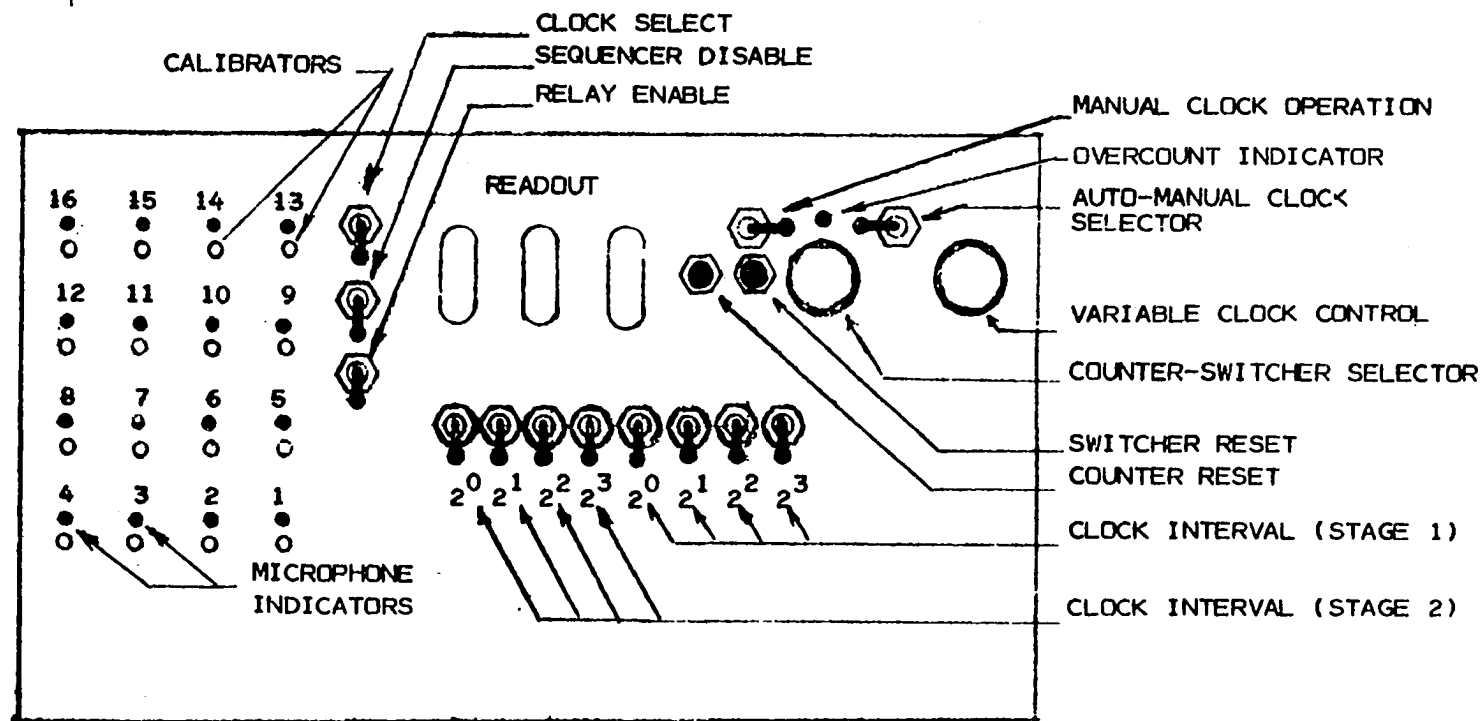
Figure 10. Back panel layout

into this connector and the other end plugs into the A3 M microphone receptacle on the sound level meter. This connection is shown on the block diagram by the line designated "signal" leading from the switching block to the sound level meter.

The direct current (DC) polarizing voltage for the condenser microphones is connected in reverse direction from the sound level meter through the utility cable to pin four of the A4 M connector on the back panel and to the input side of each DC switch. The output terminal of each DC switch returns to its respective microphone via pin two of the microphone connectors on the back panel. This arrangement eliminates cross talk between the selected microphone and the rest of the matrix. The third switch in the relays is used for purposes of calibration to correct variations in microphones and cables and will be discussed in a subsequent paragraph.

The relays are energized in sequence by circuitry in the block designated as switching logic and drivers.

Switching logic and drivers The main elements of this circuit are a four-bit binary counter and a four-line to 16-line decoder to translate the binary counts to a zero to 16 mode for sequencing the 16 microphones. For purposes of explanation, consider the number one reset block in Figure 9. Depressing the switcher reset (Figure 11) will reset the counter to binary 0000. The 16 decoder output lines identified from zero to 15, as designated in accordance with high positive logic notation, will be with output line zero at low or zero voltage, while lines one through 15 will be at high or plus five volts. Each of the 16 decoder



FRONT PANEL

Figure 11. Front panel layout

outputs are followed by an inverter stage which reverses the logic causing the zero line output to be high and the one through 15 outputs to be low. The output of each inverter drives a switching transistor for each of the 16 relays. A high output from an inverter will cause the switching transistor to energize its respective relay. When a relay is energized, a light-emitting diode indicator is also energized to provide visual indication of the microphone being used at a given time in the sequence. While it is of interest in normal monitoring situations to determine the progress of any matrix cycle, the microphone indicators (Figure 11) are especially useful when the switcher is being manually cycled during the microphone calibration procedure.

Returning to the conditions of reset as previously stated, microphone one relay will be energized with the indicator light on and the remaining relays and indicator lights will be off. Continuing under normal operating conditions from the point of reset, a positive trigger pulse from the counter-switcher block will move the counter to binary one (1000). The decoder will respond with a low at the second output line (designated 1) with the first output (designated 0) returning to high. This will energize the second microphone relay and light the number two indicator. The next timed trigger pulse will energize the relay for microphone three and this process will continue around the matrix. Since the operation of this system is dependent on precise timing pulses, it is appropriate to discuss the timing circuits used.

Timing circuits Depending on the function, three separate clocking methods were used: (1) a highly accurate crystal-controlled clock

which produces 200 kHz pulses for the analog to digital (AD) converter and command pulses 80 milliseconds in duration, the occurrence which is controllable in one-tenth second intervals from one-tenth of a second to 24 seconds, (2) a continuously variable clock which generates five millisecond duration pulses at intervals controllable from one pulse per second to 5,000 pulses per second, and (3) a manual clock which develops a pulse each time a switch is manually moved from set position to trigger position.

Crystal-controlled clock This clock consists of a one megahertz crystal-controlled oscillator followed by a series of digital dividers. The first divider stage divides by five to produce the 200 kHz clock pulses for the AD converter. A divide-by-two stage reduces the 200 kHz frequency to 100 kHz and this is followed by a succession of five divide-by-10 or decade dividers to develop the basic timing pulses occurring at one-tenth second intervals. These basic pulses are further divided by two programable divider stages, the first of which can be set to divide by any number from one to 16, and the second will divide by any number from one to 15. By selecting the proper combinations in each stage, it is possible to divide the one-tenth second basic pulses to create timing pulses occurring at intervals ranging from one-tenth of a second to 16×15 tenths of a second or 24 seconds.

The variable division in the last two stages was accomplished by four binary mode switches for each stage (Figure 11). Ordinarily these stages will count from binary 0000 to 1111 with the next count returning the stage to 0000. In decimal form this is equivalent to counting from zero

to 15 for a total of 16 counts. If, however, any of the four control switches are in the upper or on position, the stage will not return from binary 1111 to 0000, but instead will return to the number represented by the on switches. For example, if switches two and three for the first stage were on, this stage would return to binary 0110 instead of 0000. Binary 0110 is equal to decimal 2^1 plus 2^2 , or six. As a result, the stage would divide by 16 minus six, or 10. The carry output of this divider will trigger the final divider. Since this divider is the terminal divider, a count is lost and the true division will be 15 minus the number (N) preset by the programming switches. It should be noted that if either of these programable dividers has all four of its respective switches in the 1111 positions, which would cause unity division, the other divider will revert to the 15 minus N mode.

The output of this clock is a pulse 50 milliseconds in width and occurring at the programmed frequency. This final pulse is formed by a 555 timing integrated circuit used as a monostable multivibrator.

The crystal clock was used to produce the precise timing pulses for normal operation of the system.

Variable clock The variable clock is a 555 timing integrated circuit connected as an astable or free-running multivibrator. The frequency or interval control consists of a variable resistor which is part of the multivibrator time constant circuit. The off-on switch is integral with the control variable resistor and is in the open or off position when the control shaft is in the counterclockwise position.

This clock circuit was incorporated into the system to provide

timing pulses for servicing or calibrating the AD converter and also to rapidly cycle the event counter to a predetermined number.

Manual clock This clock circuit is a conventional manual trigger circuit which uses two NOR gates switched by a single-pole, double-throw switch. Movement of the switch lever from the right to left position will produce a noise-free trigger pulse. This manual capability was needed for individual calibration of each microphone.

Clock selection The clock or timing mode is dependent on the settings of three controls: (1) the clock selector control, (2) sequencer disable control, and (3) the auto-manual selector (Figure 11).

Clock selector control The clock selector control is a single-pole, double-throw switch. When the switch level is in the lower position, the output of the crystal-controlled clock is switched to the sequencer disable block, and in the upper position, the output of the variable clock is selected.

Sequencer disable control This control is a double-pole, double-throw switch wired in such a manner that in the lower position the output from the clock selector is routed directly to the auto-manual selector through the line designated count one or clock. In the upper position the selected clock output is switched to the sequencer through the line designated clock. The clock pulse is converted to a four count sequence by the sequencer, and count one is used to cycle the microphone switcher or the counter when the sequencer is enabled. This count one is routed by the sequencer disable switch back through the line designated count one or clock to the auto-manual selector control.

Auto-manual clock selector This selector is a single-pole, double-throw switch that switches the clock signals previously described, or the manual clock output to the counter-switcher selector.

Counter-switcher selector A two-pole, three-position rotary switch was used for this control. In the clockwise or number one position, the selected clock pulse is routed to the event counter. In this mode the pulses can be timed and the accuracy of the crystal-controlled clock can be determined or the system can be used as a timer.

With the switch in the middle or number two position, the clock signals are routed to the microphone switcher. This makes it possible to cycle the switcher without disturbing any reading that may have been on the counter.

The switch in the counterclockwise or number three position will connect the microphone one logic output to the counter, thus enabling the counter to tally the number of microphone matrix cycles for a given monitoring period and also provide a means to accurately determine elapsed time.

Counter This counter is a three-digit, solid-state decade counter with a range from zero to 999. The logic circuitry consists of three decade (zero to nine) binary counter integrated circuits operated in cascade. The binary number in each counter is decoded by a binary to seven segment decoder integrated circuit for each digit. Each decoder drives a seven-segment, light-emitting diode (LED) readout device. The counter reset push button, when depressed, returns the counters to binary 0000, which in turn returns the decade readout to decimal 000.

The timing or clocking signal sources and control, the microphone switcher, and counter circuits have been discussed. This discussion serves as background for further explanation of the switcher-converter, which will be made from a signal tracing standpoint.

Sound level meter The microphone switcher provides a method for automatically connecting in series one of 16 remotely located microphones to the sound level meter. The analog output of the sound level meter (SLM) is a linear direct current (DC) voltage which is proportional to the weighted sound level sensed by a particular microphone. This linear voltage appears across the terminals of the SLM readout indicator which is scaled in decibels, an unlinear logarithmic scale.

Adapting the sound level meter Adapting the sound level meter for use with this system required only slight alterations in the meter. Since the microphone of the meter was removable by unplugging, adapting the switcher system to the SLM was accomplished by the use of a utility cable with an A3 F type connector on one end to plug into the SLM in place of the microphone and an A4 F on the other end to plug into the A4 M connector located on the back of the switcher (switcher back panel layout, Figure 10). The DC signal across the terminals of the SLM readout meter was brought out through a shielded line terminated by an A4 F connector which plugs into the upper A4 M receptacle located on the back of the switcher. The full scale amplitude of this signal is .175 volts, and it must be amplified before conversion to digital form.

Linear DC amplifier This amplifier was made up of a 741 operational amplifier integrated circuit used in the noninverting mode.

Negative feedback from the output to the inverting input, as determined by an adjustable precision resistor, controls the gain of the stage. This DC amplifier is linear and the gain is adjusted so that the full scale voltage of .175 from the SLM will be amplified to approximately five volts as needed for conversion to digital form. It can be observed from the block diagram (Figure 9) that the output of the linear amplifier leads to the analog to digital (AD) converter by way of the microphone calibrator block.

Microphone calibrator The output of the linear amplifier is connected by shielded line to a common termination of one terminal of each calibrator reed switch in the 16 switcher relays. The other terminal of each reed switch leads to the input of a 220 k ohm miniature adjustable calibrator resistor. The output of the calibrator resistors are commoned and lead to the input of the AD converter, which has an input resistance of one megohm to signal ground. For a given microphone position, the combination of a 220 kilohm variable resistor in series with the one megohm input resistor to the AD converter forms an adjustable voltage divider, which makes it possible to compensate for variations in microphones and microphone lines used at the various microphone positions. The calibrators are arranged in matrix form on the left side of the switcher front panel in close proximity below the indicator light for its respective microphone. This layout, plus the manual clocking provision, facilitated calibration, so that the input to the AD converter was identical for a given sound level from one microphone position to the next irrespective of the normal variation in microphones and cables.

AD converter The purpose of this circuit was to convert to proportional digital count the output of the linear amplifier which is a voltage representation of the sound level being measured at a particular time. This converter consists of four sections: (1) linear ramp generator, (2) comparator, (3) counter, and (4) readout logic.

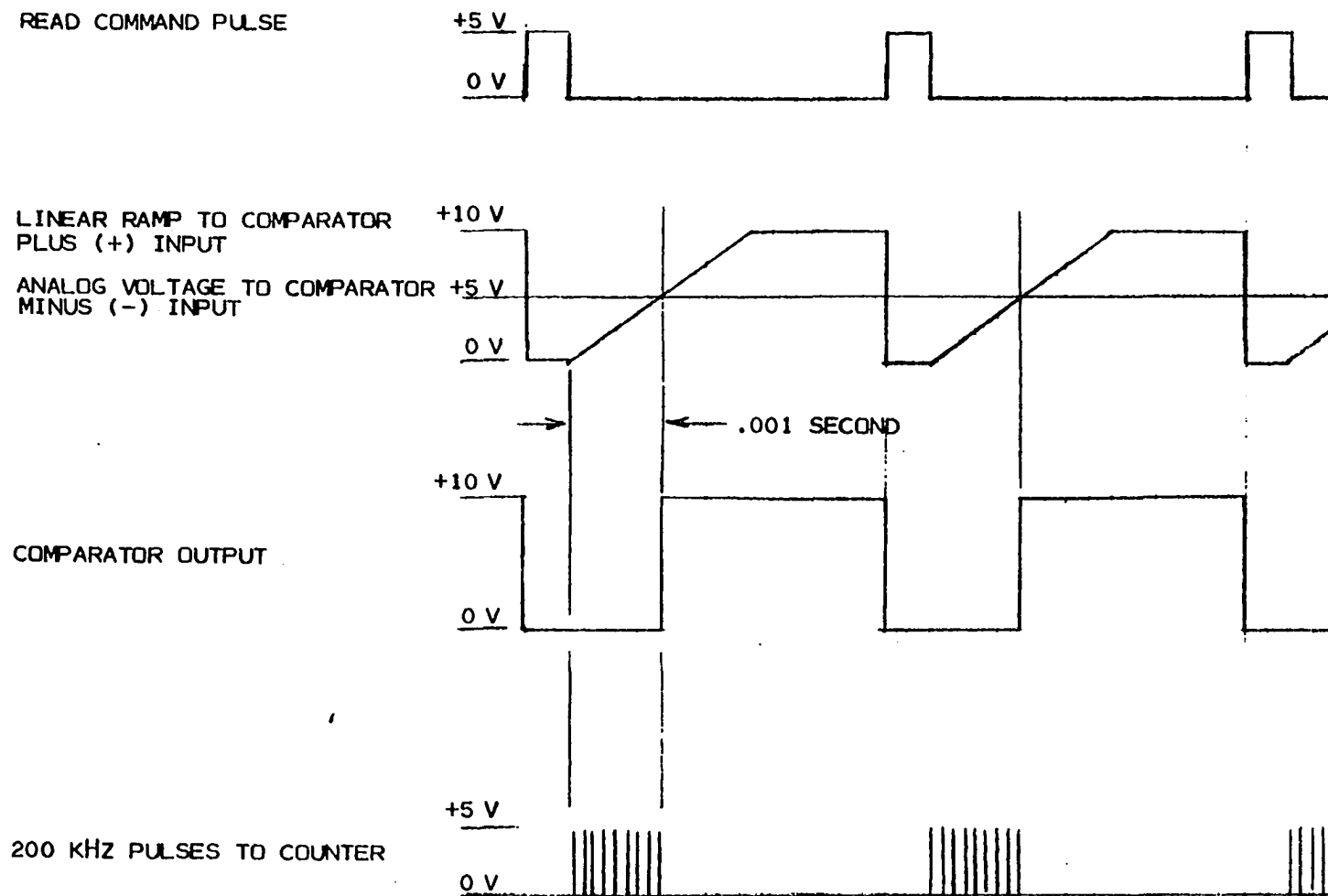
Linear ramp generator This generator produces a rising voltage or ramp which rises at a linear rate from zero volts to five volts in one millisecond (.001 second) and will continue to a maximum of 10 volts. The rising voltage is generated by charging a temperature-stable capacitor with a constant current source which results in the production of a voltage that increases in a linear manner with respect to elapsed time. The start of the ramp is triggered by the 80 millisecond system command pulse initiated by the crystal clock and timed by count four of the sequencer. This positive pulse will cause a transistor switch connected across the charge capacitor to discharge the capacitor to zero volts and hold for the duration of the command pulse. At the precise ending of the command pulse, the rising ramp will begin. This ramp voltage is impressed on the plus or noninverting input of a 311 comparator integrated circuit (IC).

Comparator A LM 311 comparator IC was used for this part of the system. This device has two inputs; one is designated as plus or noninvert and the other as minus or invert. When the plus input is at a potential which is positive with respect to the minus input, the output of the device changes from a low to a high logic position (positive voltage). This change is instantaneous, as is the change to a low or zero voltage

when the relative voltages of the inputs are reversed. In other words, this circuit compares the voltages at the two inputs and responds with either a high or a low at its output depending on the amplitude relationship of the two inputs. In this system the analog voltage from the linear amplifier is impressed on the comparator minus input and the output of the ramp generator is on the plus input. The output of the comparator controls the transistor clock pulse disable switch in the counter circuit.

Counter Since the readout of this system is in eight-level binary form on punched tape, an eight-level binary counter was required. The 200 kilo Hz clock pulses from the crystal clock are routed through a transistor clock disable switch to the counter. A logical low voltage on the control line of the disable switch will permit the clock pulses to reach the counter and counting at the 200 kilo Hz rate will take place until terminated by a logical high voltage at the clock disable switch control terminal. The output of the comparator controls the clock disable switch and also the counting period. The beginning of a counting period is determined by the ending of the read command pulse from the sequencer (count four).

This circuit can be better understood by referring to the timing diagram in Figure 12. Assume that five volts from the linear amplifier is applied on the invert or minus input of the comparator and the ramp generator output is at a high positive voltage, having had time to "run up" to a maximum of 10 volts. The output of the comparator is high and no 200 kHz pulses are being counted. With the arrival of the read command pulse, several operations take place: (a) the linear ramp is



CONVERTER TIMING DIAGRAM

Figure 12. Converter timing diagram

brought to zero and this is reflected on the plus input of the comparator; (b) the five-volt analog voltage on the minus input to the comparator causes the comparator output to drop to logical low or zero volts; (c) a diode switch instantaneously switches the positive command pulse voltage to the clock disable switch to prevent counting from beginning until the end or trailing edge of the command pulse, as is needed for precise timing with this system; and (d) the reset input of the counter is connected to the command pulse line and this positive pulse resets and holds the counter in this state until the end of the command pulse.

When the trailing edge of the command pulse drops to zero, the timing function begins with three simultaneous happenings: (1) the counter is no longer held at reset; (2) the clock disable switch is deactivated to permit counting of the 200 kHz clock pulses; and (3) the output of the ramp generator begins to rise.

In the example shown in the timing diagram (Figure 12), the linear ramp will rise until it equals the analog voltage. At this point the comparator output will rise and stop the count. The elapsed time for this event was one millisecond or 200 counts. The binary equivalent of decimal 200 is 11001000 or 2^7 plus 2^6 plus 2^3 and 128 plus 64 plus 8 equals 200. This number will remain in the counter to be read out during the punch cycle.

Readout logic The purpose of the readout logic circuits was to sense the logic states (high--plus five volts, or low--zero volts) at each level output of the counter, and to transfer this information in the proper form to the tape punch so that a high in the counter at a given

level will cause a hole to be punched in the track corresponding to this level on the paper tape. Figure 13 shows an example of the punched tape format.

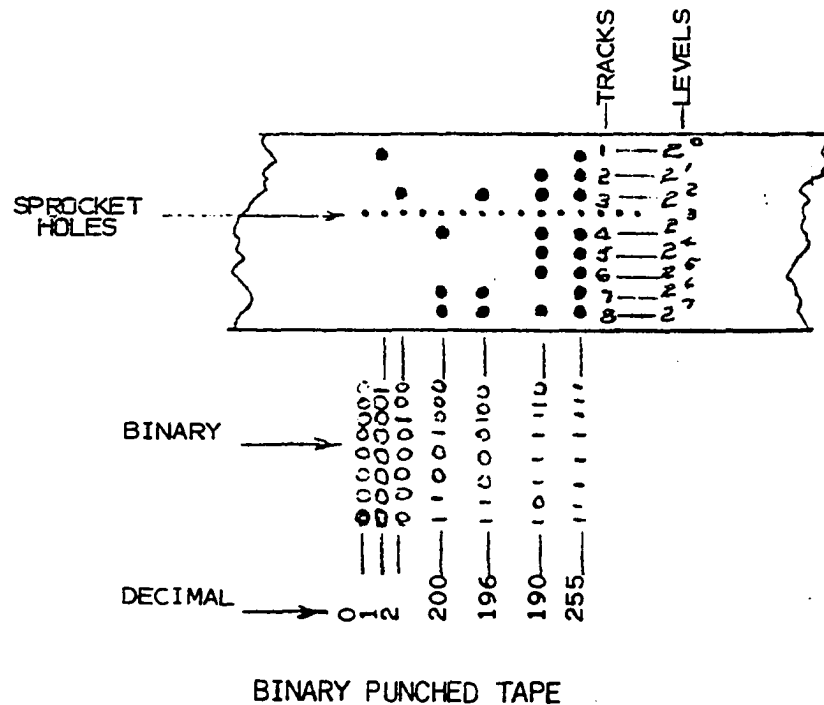


Figure 13. Binary punched tape

The logic approach to this readout system was based on nine two-input NAND gates, one for each binary level in the counter and one for the punch clutch mechanism. The readout function is initiated by the same count one command pulse from the sequencer that was used to cycle the microphone switcher. This pulse is simultaneously applied to one input of each NAND gate for each binary level in the counter and to both inputs of the clutch NAND gate. The output for each binary level in the counter is connected to the other input of a NAND gate for a particular level. It is characteristic of a NAND gate to respond with a low at the output only when both inputs are high and this is the key to the readout

function. Again consider a count of 200 and the punched tape response for this number as shown in Figure 13. This count is represented by a punched hole in track eight, track seven, and track four. Also for this number, the outputs for levels 2^7 , 2^6 , and 2^4 will be high and will appear at one input of the NAND gate that corresponds to each of these levels. Under these conditions, the punch command pulse (count one from the sequencer) will cause the output of these three binary level gates and the clutch gate to become low, as is required to operate the punch. The outputs of the readout NAND gates connect through shielded lines to nine pins of a 24-pin plug which is accommodated by a similar socket mounted on the punch chassis. The punch was designed to operate on low logic to guard against punch runaway in case the logic lines should become unplugged while the system was energized.

Count limiter The accuracy with which the computer was able to read and process the data on the punched tape was dependent on the total count being limited to 240. This was accomplished by connecting the outputs of the upper four levels of the counter (2^7 , 2^6 , 2^5 , and 2^4) to one section of a dual four-input NAND gate. When all of these levels become high for a count of 240, the output of the gate returns to low. This output is routed to one input of the remaining section, while the other three inputs are permanently held low. In this mode, the second gate acts as an inverter with the output going high when count 240 is reached. This high is switched by an isolation diode to the clock disable switch to terminate the count. If count limiting occurs, a warning light on the front panel remains on until manually reset. This serves as

a warning to change the SLM range setting and check the tape for the number of 240 counts.

Throughout this discussion, the need for command pulses at precise cyclic intervals has been emphasized. This function is performed by a system module known as the sequencer.

The sequencer When this system is monitoring sound from a microphone position, a series of repetitive operations is required: (1) the next microphone in the matrix sequence must be selected and the data from the last microphone position punched on tape; (2) a logic test must be made after switching to the next microphone position to determine if it is microphone number one in the sequence; (3) a sufficient amount of time at each microphone position must be allowed for the SLM to accurately monitor the sound; and (4) the analog output of the SLM must be converted to digital form for readout. The sequencer circuitry for performing this repetitive cycle was comprised of three sections: (1) a counter, (2) a decoder-inverter, and (3) count logic.

Sequencer counter A four-bit binary counter IC, identical to the one used for the microphone switcher, performs the counting function from zero through three for a total of four counts. At reset, the binary output levels will be at binary 0000. The first programmed clock pulse after reset will change the counter to 1000, which will be followed by 0100 and finally to 1100. On the next or fourth clock pulse, the counter will move to 0010, but is reset to 0000 for a total of four counts.

Sequencer decoder Binary to decimal decoding was accomplished by a four-line to 10-line decoder. The outputs are designated from zero to nine with binary 0000 in causing the zero line to be low and the remaining line to be high. These outputs are inverted to be compatible with the count logic gates that follow.

Sequencer count logic The count logic consists of four two-input AND gates. In contrast to NAND gates, which were used for the read-out logic, AND gates respond with a high level output when both inputs are high at the same time. One input for each AND gate is connected to the clock line while the inverted output from decoder lines zero, one, three, and four lead to the second input of an associated AND gate. The outputs of these AND gates are identified in sequential order as count one, count two, and count four. It should be pointed out that there is no AND gate for decoder line two and the fifth AND gate for count five is used to reset the counter to 0000. This reset is for all practical purposes instantaneous as determined by the propagation delay of the gates involved.

As a means of clarification in regard to the operation of the sequencer, consider Figure 14 and the operating cycle starting with reset clock pulse. The leading edge of this 80 millisecond positive pulse will shift the counter to binary 0010 and is simultaneously impressed on one input of each of the sequencer AND gates. This will cause the number four output of the decoder to be at zero or low. This pulse is inverted bringing one of the two inputs of the reset AND gate to a high potential. The clock pulse being at the other input causes a high to appear at the output of reset gate four. Note that this pulse is of short duration,

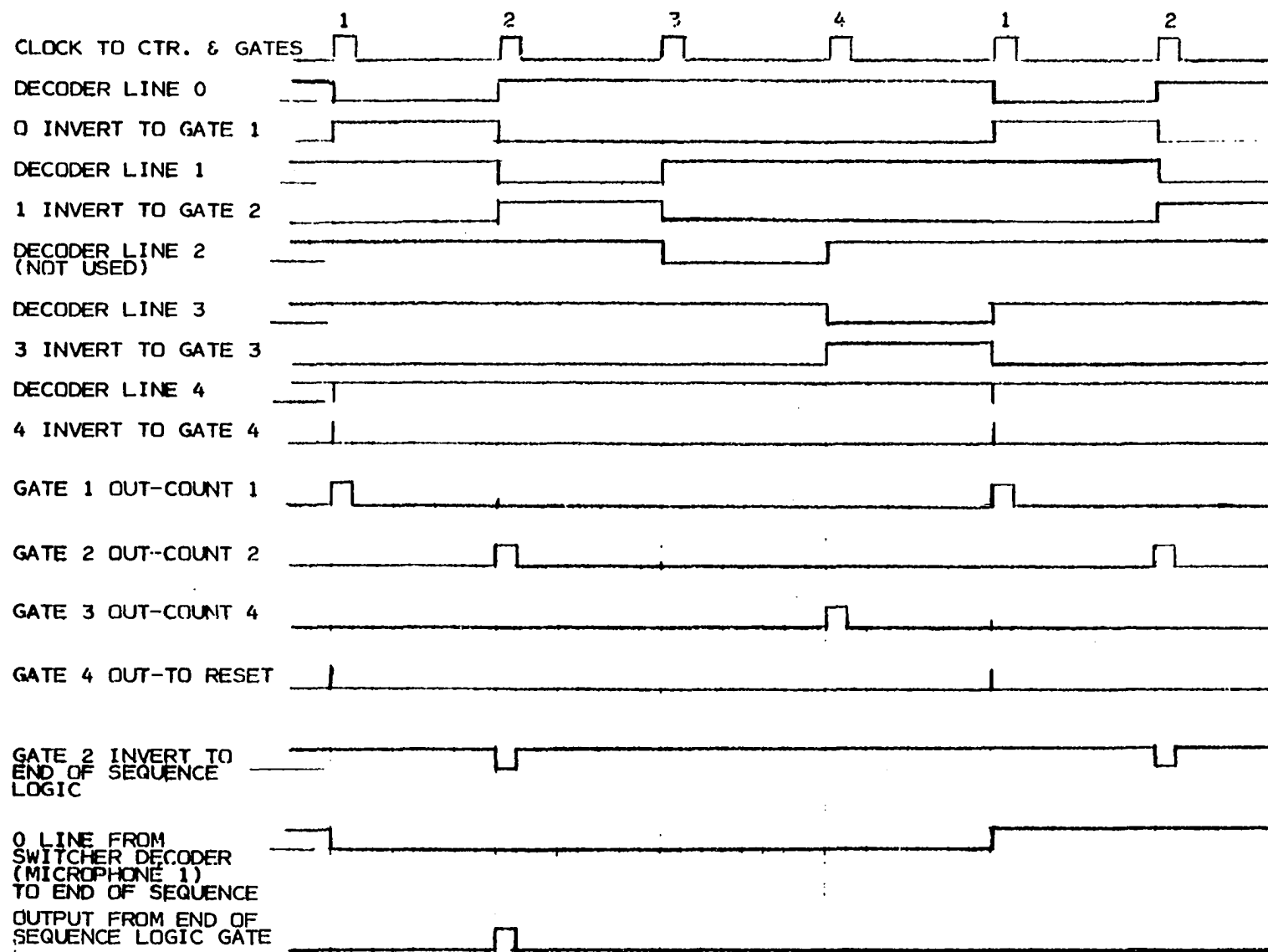


Figure 14. Sequencer timing diagram

for as soon as the counter resets, this gate will have only the clock pulse remaining at one of its inputs and the output goes low. This reset switching occurs in two microseconds and is determined by the switching time inherent with these gates and the counter. When the counter returns to reset, the inverted output of decoder line zero impresses a high on one input of the count one gate which, in conjunction with the clock pulse remaining on the other input, causes the gate one output to produce the positive count one command pulse. This pulse is responsible for switching the microphone switcher and initiating the punch cycle. Although the inverted decoder output will be present for an entire count, the output at the count one gate will last only as long as the clock pulse. It is in this manner that one clock pulse source can be used to produce identical clock pulses at all sequencer outputs.

The next clock pulse will cycle the counter one step and, as previously explained, the output of the count two gate will be positive for the duration of the clock pulse. This count two output is inverted and connected by a shielded line through a six-pin disconnecter to the punch. The line zero output for the microphone switcher decoder is brought through the same plug to the punch. The low or inverted output from the count two gate and the low from the microphone one decoder combine to produce a high out from the end-of-sequence logic gate (Figure 14). This in turn will trigger binary 11111111 end-of-sequence punch which identifies the completion of a microphone matrix cycle. It should be noted that the end-of-sequence logic is the opposite of the NAND logic used for the readout discussed previously. This is accomplished with a two-input

NOR gate which responds with a high out when both inputs are a zero or low. The use of this type of logic for the punch commands was used to protect the punch from entering a runaway mode if a control line should open or a power supply fail in the switcher circuitry.

The third count will cycle the counter one step and allow one additional count for the SLM monitoring time which began when the microphone was switched by the first count.

Count four from the clock cycles the counter one step. The output of gate three will be high to initiate the count four read function. The AD converter will convert the analog signal from the SLM to digital form in one millisecond and store this information for the punch cycle occurring on the next clock count.

The clock count that follows will bring about the reset or count one step to begin the repeat of the sequencer cycle.

Design and fabrication of the punch unit

As can be noted from the block diagram in Figure 9, the punch unit consists of a motor-driven, relay-controlled, eight-level paper tape punch, the punch code logic and keyboard transfer circuits, the end-of-sequence logic circuits, the DC power supplies, and control switches.

Tape punch A standard teletype punch was modified so that it could be driven with a miniature V-belt instead of the conventional gear drive. In addition, the punch selector relays and clutch relays were reworked to be compatible with transistor switching in lieu of mechanical switching. The punch and its companion drive motor were mounted on a seven- by 14-inch frame which also accommodated the power supplies used

with the system. When the punch is in operation, the clutch input member rotates continuously. A punch command will cause the clutch relay to unlatch the clutch and the mechanism will rotate one revolution to complete the punch function. The eight punches, one for each binary level, are selected according to the input logic and triggered by the punch command along with the clutch. The combination of punches is selected as follows: (a) the punch command will activate the clutch and the selected punch relays; (b) activating a punch relay will release a latch which will cause a punch to be forced through the paper by a cam actuator during the single rotation of the punch mechanism.

The punch relays and the clutch relay are energized by an 80 volt DC power supply. One side of each relay coil is connected to the power supply and the other leads to a switching transistor through a 330 ohm current limiting resistor. Each switching transistor is driven by a driver stage through a 1000 ohm resistor. A low or zero voltage on the input of the driver transistor results in a low to the input of the switcher transistor and no current will flow through the transistor. A high potential from the punch logic gate will cause the switching transistor to act as a closed switch, thus causing the relay to be energized. The diode across the relay coil serves to dampen the inductive current surges developed by the relay coil when being switched from on to off.

Punch logic circuit The punch unit was designed to operate on low or zero logic inputs in order to prevent damage to the punch mechanism in case of an open control line or loss of control power. In addition, this punch unit was designed to be operated from an ASCII

keyboard encoder with provision for selecting either the encoder or the normal monitoring output. This dual capability was accomplished through the use of nine, two-input NOR gates, one for each punch relay and the clutch relay. The characteristics of a NOR gate are such that if both inputs are simultaneously low, a high will occur at the output; therefore, with one of the inputs held at low, the output will be inverted from whatever logic state is present on the other input. In this system the nine logic lines (eight punch lines and the clutch line) from the keyboard encoder lead to one input of its respective NOR gate while identical lines from the switcher lead to the other inputs. It is now apparent that in order to operate from one input or the other, the input lines of the opposite unit must be held at a low potential. This requires multiple switching of nine independent lines from one common switching control. Also the nine logic lines from either the keyboard encoder or the switcher must not be grounded when that unit is operational or the logic out circuitry may be damaged. Switching or keyboard transfer was accomplished through the use of three hex inverter integrated circuits. Each hex inverter consists of six independent amplifiers, for a total of 18, which were needed for this switching operation. An inverter is a type of amplifier which will have a high or open switch characteristic from its output to ground when the input is low or at ground potential, and when the inputs are ungrounded or high, the outputs will be switched to ground or low, an action identical to the opening and closing of a switch. By connecting an inverter output to each of the nine NOR gate inputs for the switcher and the remaining inverter outputs to the NOR

gate inputs for the keyboard encoder, these two groups of nine logic lines can be independently switched by grounding or ungrounding two separate control lines. One control line connects to the nine inverter inputs for the switcher and the other to the nine inverter inputs for the keyboard encoder. These control lines are grounded or ungrounded in correct relationship to the switcher and keyboard power off-on switches by cross wiring of the proper contacts of the double-pole, double-throw power switches. This system made it possible to selectively ground the output logic lines of the unit not in use while the other was performing in a normal manner. If for some reason both the switcher and keyboard encoder power switches were switched on, the punch would function only if a keyboard key was depressed during a punch command pulse from the switcher. This would be an obviously invalid operation.

End-of-sequence punch This punch, which occurs at the completion of each microphone sequence, is initiated by an inverted or low count-two pulse from the sequencer and a low from the microphone switcher decoder for microphone one. The simultaneous occurrence of these low pulses at the inputs of the end-of-sequence logic NOR gate will cause the output of this gate to be high, thus completing the logic test to determine if microphone one which follows microphone 16 is energized. This high output will activate a transistor driver, and this in turn activates a heavy duty switching transistor which is connected through an isolation diode to the clutch relay and to each of the level punch relays through an isolation diode in series with a miniature plug-type switch. Manually setting these switches enables the operator to select any desired end-of-

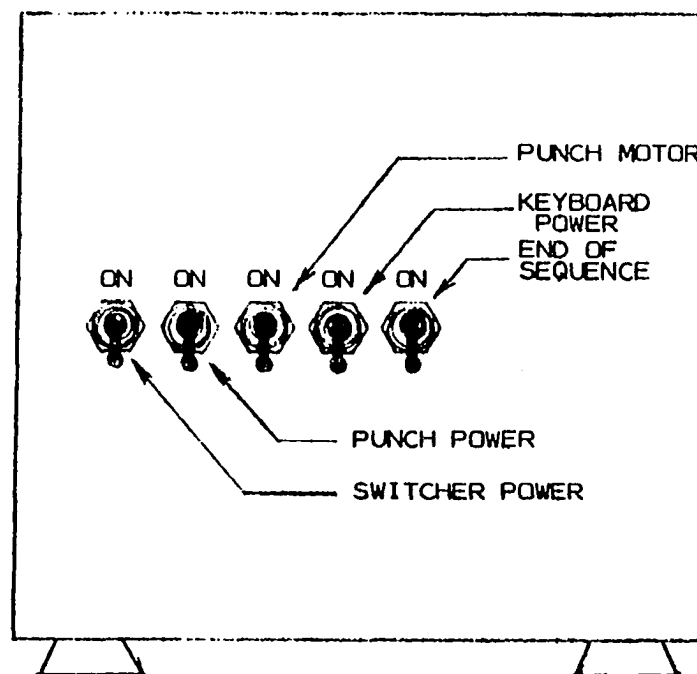
sequence punch (11111111 was used for this project). When activated, the heavy duty switching transistor will cause current to flow from the 80-volt supply through the isolation diodes and any closed punch relay switch to produce the desired punch code. The isolation diodes allow current to pass only one way and will eliminate interaction between the end-of-sequence driver and the data drive transistors.

Power supplies The location of the control switches are shown in Figure 15. From left to right, the switches are: (1) the power switch for the microphone switcher circuitry, (2) the punch power switch, (3) the punch motor off-on, (4) the keyboard encoder power switch, and (5) the end-of-sequence off-on switch.

The plugs and fuses located on the punch unit back panel are shown in Figure 16. The two switches are power line voltage compensation switches for the plus 80-volt punch relay power supply. The effect of the various switch combinations in reference to 117 volts input is as follows: combination one and four reduces the output by six percent; positions two and four give a nominal or zero change; the combination of one and three raises the voltage by six percent; and switch position two and three raises the output by 12 percent.

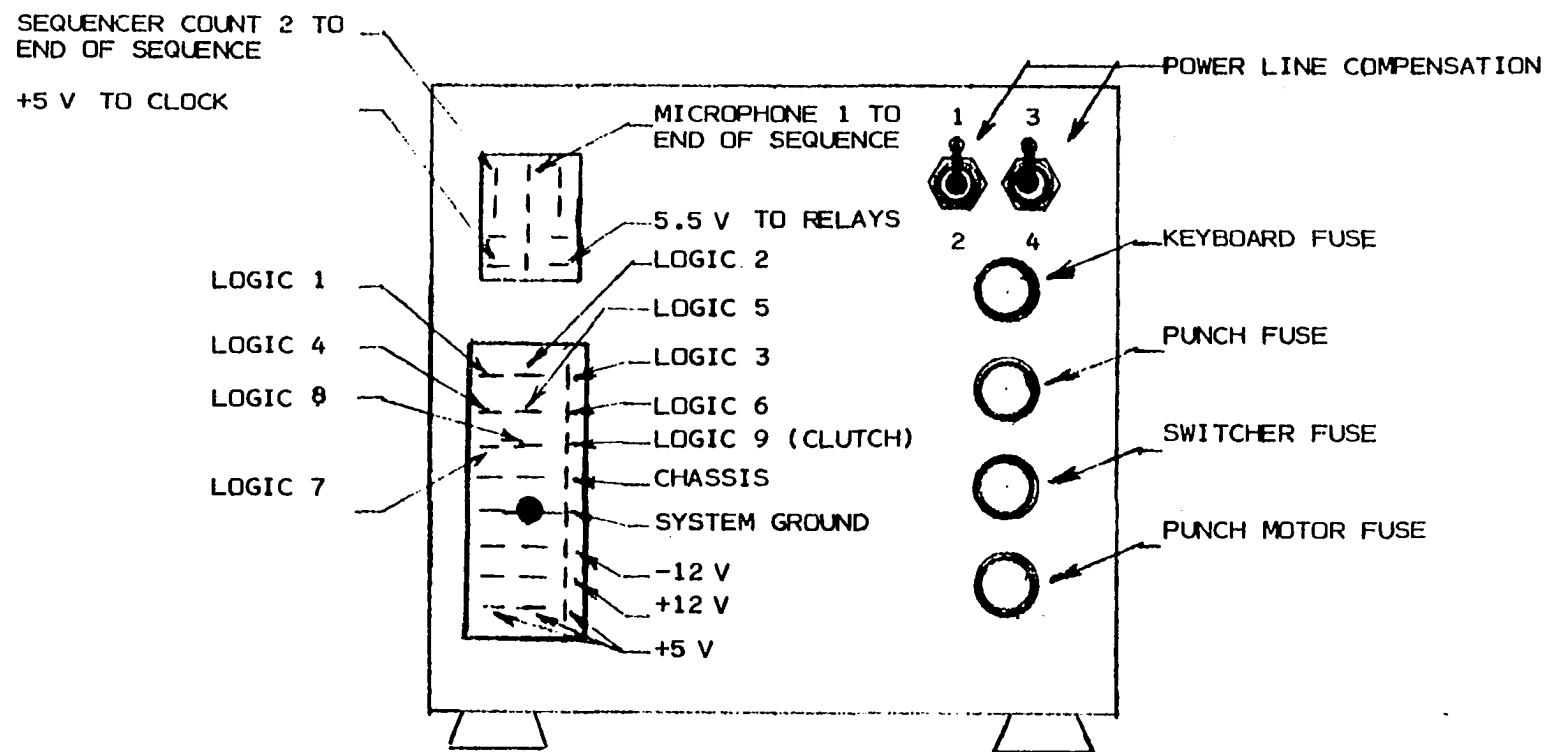
Testing and Initial Calibration of the Instrumentation

The measurement standard for this instrumentation system was an ANSI-approved Type 2 sound level meter (SLM). The readout device for this type of instrument is an analog meter calibrated in an unlinear decibel scale. The input to this readout meter is a low-level, direct current (DC) signal with amplitude variations being in proportion to monitored



PUNCH UNIT FRONT

Figure 15. Punch unit front



PUNCH UNIT REAR

Figure 16. Punch unit rear

sound levels. In order to adapt the automated instrumentation to the Simpson 886 SLM, it was necessary to establish the relationship between the physical movement of SLM readout meter indicator and the input voltage. As is characteristic of this type of meter, the physical movement of the indicator will be linearly proportional to the input voltage. The reading in decibels (dB) will not vary in linear proportion because the decibel scale is based on logarithmic or exponential change.

The readout of the Simpson 886 SLM is divided into nine regular ranges from 50 to 130 dB and a special OSHA range. The indicator meter dial is calibrated in one dB increments from minus 10 dB to plus 10 dB for the regular ranges and from 85 to 115 dB for the OSHA range. When using the instrument on any of the regular range settings, the output reading is equal to the range setting plus the dial indication. For example, a range setting of 90 dB and an indication of minus five dB would correspond to a reading of 85 dB.

The relationship between the readout meter input voltage and the indicator deflection was established by exposing the SLM microphone to a variable amplitude 1000 Hz sound source which made it possible to bring the readout indicator hand to any desired reading while measuring the input voltage with a laboratory-type digital voltmeter. At full physical deflection or a reading of 10 dB, the analog input was .175 volts. A physical half-scale deflection or four dB reading required .0875 volts, which is exactly one-half the full scale input voltage. For a reading of zero dB, which is 31.6 percent of physical full scale, the input voltage was .0553 volts, and at minus 10 dB or 10 percent of physical full scale

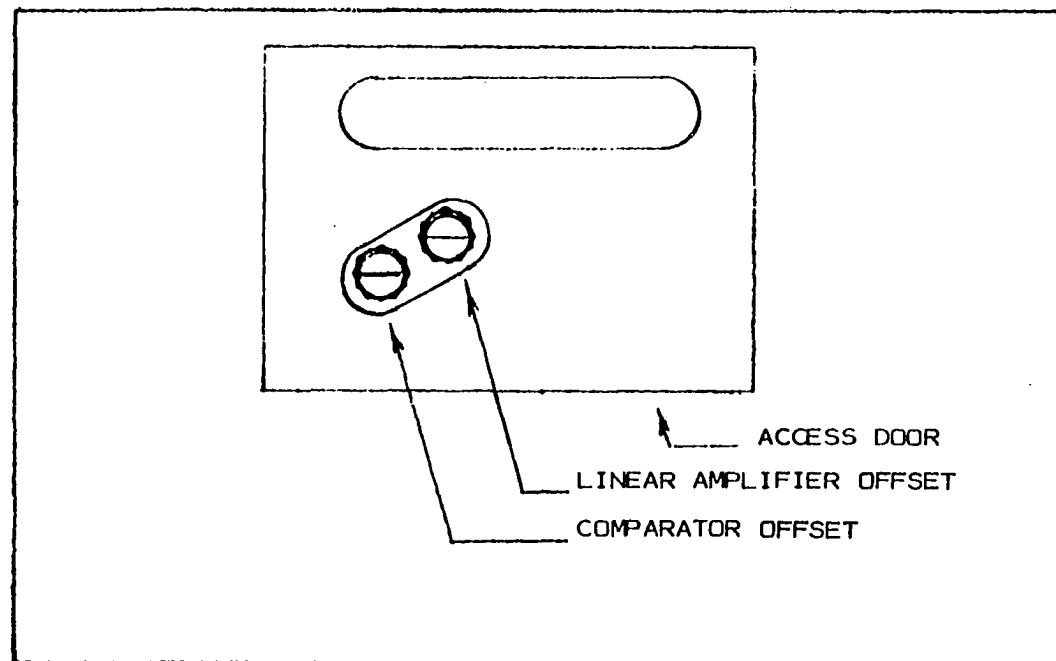
the voltage was .0175 volts. The consistency of the linear relationship between physical indicator deflection and input voltage to the readout meter was confirmed by these readings and established the validity of using this voltage as the interfacing medium between the SLM and the automated instrumentation.

The voltage appearing across the input of the SLM readout meter was conducted through a shielded line to the input of the linear direct current (DC) amplifier. This amplifier consists of a selected operational amplifier integrated circuit (IC) used in the noninvert mode. The precision gain control potentiometer was adjusted for a linear gain of 28.57 times so that the full scale input voltage of .175 volts was amplified to five volts DC. This control was located on the back panel (Figure 10). This amplifier was found to be accurate and all input voltages were proportionally amplified. The input impedance of this amplifier was of such magnitude that the readings of the SLM meter were not altered in a measurable manner. The output of the linear amplifier passes through the microphone calibrators to the analog to digital (AD) converter input.

The purpose of the AD converter was to change the amplified analog signal from the linear amplifier to digital count. The initial calibration of this circuit was performed with a four-trace oscilloscope to monitor the various timing functions while the converter was clocked by the built-in variable clock set at a frequency that would give clear oscilloscope traces with a minimum of flicker. The linear ramp was adjusted to rise from zero to five volts in one millisecond, which would permit 200 counts from the 200 kHz clock for the counter. Access to this

adjustment was provided on the back panel (Figure 10). It is important to note that the linear ramp does not stop at five volts, but continues to a final level of 10 volts during each reading. This extended range of the linear ramp provides ample extension for a count of 240, or a count of 40 over the 200 count set for a full-scale reading of plus 10 dB. This overcount extends the range to one and six-tenths dB above the full scale reading of 10 dB. The zero-input, zero-count point was achieved through the adjustment of the linear amplifier offset balance control and the comparator offset balance control, the location of which is shown in Figure 17. These adjustments compensate for variations in integrated circuits. Initial adjustments were made by varying the balance control for the linear amplifier so that a zero input voltage results in zero output voltage. The balance control of the comparator was then adjusted to give zero count. After making these initial control settings, slight final adjustments of the linear ramp generator adjustment, the linear amplifier gain control, and the linear amplifier offset adjustment made it possible to achieve the desired linear count relationship between input signal and the output counts. After preliminary trials, full scale (plus 10 dB) readings on the SLM brought about consistently accurate counts of 200, with counts of 100 for half scale (plus 4 dB) and 20 for one-tenth scale (minus 10 dB). The zero dB reading, which is 31.6 percent of the physical full scale, resulted in a count of 63 to satisfactorily correspond with the theoretical count of 63.24.

Because of the unlinear characteristics of the logarithmic dB scale, the counts per dB ratio is not equal throughout the scale. While the



SWITCHER TOP VIEW

Figure 17. Switcher top view

average is 20 counts per dB, the ratio from 190 to 200 counts is 22.4 counts per dB, and for the 20 to 30 count range, 2.84 counts per dB.

Collection of Data

The testing of the instrumentation was conducted in the wood laboratory of the Industrial Education Department at Iowa State University. As can be noted from the laboratory layout (Figure 18), this laboratory is equipped with a variety of machines which served as sound sources emitting from diverse locations within the laboratory. The response of the equipment was observed by monitoring a series of treatments which consisted of operating the machines singly and in combination.

The microphone matrix placement (Figure 18) was implemented by the use of microphone support clamps developed for this system. An extension tool was designed to permit placement and removal of the clamps without the need for ladders or similar aids. In this laboratory the light fixtures provided satisfactory support for hanging the microphones in a vertical position five feet from the floor.

The microphones used for this system were of the random incidence type and were designed to be positioned with a center axis plane at an angle of 70 degrees with respect to a line from the microphone to the sound source. With the microphones hanging vertically, this response pattern permitted reception from all directions to give uniform and repeatable results.

Because of the variations in microphones and the length of microphone cables, each microphone was individually calibrated to a standard

A	SHAPER	J	SANDER
B	ROCKWELL TABLE SAW	K	OLIVER TABLE SAW
C	LATHE	L	BANDSAW
D	SANDER	M	JIGSAW
E	MORTISER	N	RADIAL ARM SAW
F	JOINTER	O	SANDER
G	PLANER	P	SOUTH HEATER
H	UNIPLANE	Q	NORTH HEATER
I	GRINDER	R	DRILL PRESS

Figure 18. Wood laboratory floor plan

114 dB calibrator, which was moved to each microphone position as the sequencer was manually cycled from one microphone position to another. The SLM was set on the 114 dB range and each microphone calibration control was adjusted to give an output count of 100 on the punched tape.

After calibration, the SLM was removed from the system and used to monitor test treatments for comparison with those recorded by the complete instrumentation. The readings indicated that the switcher circuitry did not affect the performance of the SLM.

With the system now operational, a treatment schedule was developed, and during this process several rechecks with the calibrator at all microphones gave consistent readout counts of 100 plus or minus one count, indicating that the system was stable. The treatment schedule is shown in Table 3. As can be observed from the laboratory layout in Figure 18, many of the treatments involved machines which are separated as far as possible in the laboratory to create spatially contrasting sound sources. Other treatment combinations, such as treatments 19 and 20, were intended to create sound patterns which were concentrated in one direction or another across the laboratory.

Although this instrumentation was designed for monitoring laboratory sound levels for extended periods of time, it was adaptable to the interruptions that were necessary for changing the treatments listed on the schedule. The operating procedure used for this study was as follows: (a) unplug and remove the keyboard from the system; (b) check all plugs and cables and place the line compensation switches in the upper positions (Figure 16); (c) using Figure 15 as a reference, turn on the

Table 3. Treatment schedule

Treatment	Matrix cycle	Noise source
1	1- 4	Heaters
2	5- 8	Shaper
3	9-12	Mortiser
4	13-16	Radial arm saw
5	17-20	Dust collector
6	21-24	Jointer
7	25-28	Surfacer
8	29-32	Uniplane
9	33-36	Bandsaw
10	37-40	Jigsaw
11	41-44	Jigsaw and bandsaw
12	45-48	Jigsaw, bandsaw, and uniplane
13	49-52	Uniplane, surfacer, and jointer
14	53-56	Sanders and grinders
15	57-60	Lathe
16	61-64	Rockwell table saw
17	65-68	Oliver table saw
18	69-72	Mortiser, shaper, and radial arm saw
19	73-76	Lathe, Rockwell table saw, and Oliver table saw
20	77-80	Jigsaw, bandsaw, sanders, grinder, uniplane, surfacer, and jointer
21	81-84	Machines of matrix cycles 69-72, 73-76, 77-80, drill press, and vertical sander
22	85-88	Machines of matrix cycle 81-84 and dust collector
23	89-92	Machines of matrix cycle 85-88 and heaters

punch motor, the switcher power, the punch power, the keyboard power, and the end-of-sequence logic remains turned off; (d) referring to Figure 11, set the counter-switcher selector in position two, the variable clock control counterclockwise, the auto-manual clock selector to the right or auto position, and the relay enable in the up or on position; (e) select a one-second primary clock interval by placing all of the stage one switches in the up position except the 2^2 switch and all the stage two switches in the up position except the 2^1 switch; and (f) depress the

counter and switcher reset switches.

The counter now read 000 and the microphone indicator lights changed every four seconds from number one to two and on around the matrix. In this mode, there was no punch operation because the end-of-sequence logic switch was off, and with the keyboard and switcher power switches on, the keyboard transfer logic was inhibiting all commands from the switcher unit. With the end-of-sequence logic switch on, the end-of-sequence punch (binary 11111111) was executed on the second sequencer count after the system switched to microphone one. Placing the keyboard power switch in the off position now allowed the normal transfer of information from the switcher unit to the punch unit, and a punch cycle occurred at four-second intervals along with the end-of-sequence punch each matrix cycle. Since the SLM was off, the regular punches were binary 00000000.

The system was now operational and the monitoring procedure was implemented as follows: (a) returned the SLM range to the proper setting (80 dB was used for this study); (b) prepared the machine or machines for the first treatment; (c) the system was now monitoring, but no readout punches occurred; (d) allowed the microphone switcher to move past microphone one and then switched on the end-of-sequence logic; (e) watched the microphone indicator and the instant it changed to the next microphone (any microphone except number one), pressed the switcher reset, and at the time of the end-of-sequence punch (one second after reset), switched off the keyboard power switch; (f) turned the counter-switcher selector to position three (counterclockwise) to allow the counter to register the matrix cycles; (g) after the counter indicated the completion of the

third matrix cycle for a given treatment, switched the end-of-sequence logic off; (h) observed the progress of the fourth matrix cycle, and at the moment the counter indicated the completion of this cycle, switched the keyboard power switch on; (i) turned the counter-switcher selector to position two to prevent the indicated matrix cycle from being changed while the next treatment setup was being made; (j) repeated steps d through i until the end of the schedule.

One important consideration when performing interrupted programming was to always begin a matrix cycle with an end-of-sequence punch and end all punching at the completion of recording microphone number 16 data. This format made it possible to utilize machine timing to maintain perfect matrix cycles. As a session proceeded, it was desirable to periodically check calibration. The calibration procedure was as follows: (a) at the end of a selected treatment segment, switched off the end-of-sequence logic and placed the keyboard power switch in the on position; (b) placed the calibrator in position at the selected microphone; (c) moved the auto-manual clock selector to the left or manual position while depressing the switcher reset; (d) moving the manual clock operation, switching from left to right stepped the microphone switcher to the desired position as indicated by the microphone indicator light; (e) returned the keyboard power switch to off; (f) set the SLM on the 110 dB range; and (g) energized the calibrator.

The calibrator level was now monitoring by the selected microphone and the level punched continuously on the tape. Any deviation from a count of 100 was corrected by adjusting the microphone calibrator for

that position. After performing the calibration, the system was returned to monitor readiness by returning the keyboard power switch to off and the auto-manual clock selector to auto (right) position.

At completion of the treatment schedule, 1472 data readings were punched in the paper tape to be read directly into the computer for data processing.

The data in binary form was read into the computer and converted to decimal counts. These results were stored in computer memory in row-column format with each column representing the readings for the microphone positions and the rows corresponding to matrix cycles. The computer was then programmed to convert these decimal counts to decibel readings and were printed in the row-column format (Figure 19). The conversion to decibels was achieved through the use of the following equation:

$$\text{dB} = 20 \log \frac{\text{observed count}}{100} + \text{SLM dB range} + 4.$$

For example, assume the SLM is set on the 80 dB range and the observed count was 165. It was explained in previous text that a physical mid-range reading of the SLM readout meter occurred at count 100 and that the dB midscale reading was plus four dB. Using count 100 or 84 dB as a reference, the logarithmic ratio between count 100 and the observed count can be calculated. In this example, the equation is:

$$\text{dB} = 20 \log \frac{165}{100} + 84 = 88.35 \text{ dB},$$

This discussion has presented the design, function, and operation of the sound monitoring system. The results of 23 machine combinations or

	MICROPHONE POSITIONS												
	1	2	3	4	5	6	7	8	9	10	11	12	13
	63.085	63.085	62.062	64.000	63.085	64.000	62.062	68.082	64.000	63.085	59.563	68.609	60.902
	63.085	62.062	62.062	63.085	62.062	62.062	62.062	68.082	65.584	64.828	63.085	70.021	64.828
	63.085	63.085	64.000	64.000	62.062	64.828	62.062	68.609	66.279	64.828	64.828	70.848	66.279
1	64.828	65.584	63.085	64.828	62.062	67.522	64.828	68.609	68.609	66.279	64.000	70.021	65.584
	81.025	80.258	73.542	71.959	77.064	76.041	75.126	72.627	75.126	74.630	72.943	71.959	70.021
	80.258	80.258	73.542	72.627	77.255	76.256	74.103	71.604	74.103	72.943	71.959	72.299	70.444
	80.650	79.707	72.627	71.559	77.064	75.126	74.370	71.959	74.881	73.248	71.235	70.848	69.105
2	79.848	79.848	73.827	71.235	77.064	75.821	74.103	70.021	73.827	73.542	70.848	71.604	70.021
	59.563	60.902	64.828	70.444	62.062	62.062	65.584	67.522	57.979	59.563	57.979	60.902	53.542
	60.902	62.062	64.828	71.235	60.902	59.563	62.062	66.279	56.041	56.041	59.563	60.902	53.542
	62.062	62.062	65.584	73.827	57.979	57.979	63.085	65.584	53.542	57.979	59.563	64.828	57.979
3	60.902	63.085	66.279	69.105	60.902	59.563	67.522	65.584	57.979	57.979	60.902	63.085	56.041
	64.000	64.828	64.000	62.062	70.021	68.082	64.828	64.000	75.596	70.444	69.105	68.082	73.248
	62.062	65.584	64.828	64.000	71.235	68.609	65.584	64.000	75.821	68.609	68.082	65.584	73.248
	64.828	66.923	64.828	64.000	70.444	68.082	64.828	64.000	76.041	70.021	69.105	66.923	72.943
4	64.000	64.828	64.000	64.828	70.444	69.575	65.584	64.000	75.126	70.021	67.522	65.584	73.248
	77.255	77.442	75.596	75.364	75.126	75.821	73.542	74.370	74.370	73.542	73.248	73.827	72.943
	76.256	77.442	75.364	75.126	75.364	74.630	73.827	74.103	75.364	74.370	73.248	73.542	72.299
	76.465	77.064	74.881	75.596	75.596	75.364	74.103	73.542	73.827	73.542	71.959	73.248	71.959
5	76.041	77.804	75.821	75.126	75.596	75.821	73.542	74.103	74.370	73.542	72.299	73.542	72.299
	71.959	74.103	71.604	71.604	81.025	73.248	72.627	72.943	78.648	74.630	70.848	71.235	72.627
	73.248	73.542	72.943	71.604	80.391	73.827	72.627	74.630	78.807	74.630	71.959	71.959	74.881
	72.943	73.827	71.604	70.021	80.650	71.959	72.627	73.248	77.442	74.630	71.959	72.627	74.630
6	71.235	74.370	70.948	71.559	79.987	71.604	72.299	73.542	78.964	74.630	70.444	73.248	73.248
	75.821	76.041	72.299	72.299	78.807	78.486	73.827	71.604	76.869	76.465	71.604	74.103	71.604
	76.256	76.256	72.299	70.848	78.807	78.648	72.943	70.444	76.869	76.465	71.235	74.103	71.604
	76.669	76.256	72.299	71.959	78.320	78.320	72.943	70.021	76.669	76.041	69.575	73.248	71.235
7	76.869	76.041	72.299	71.959	79.117	78.486	73.248	69.575	76.465	76.041	70.444	73.248	71.235
	75.821	80.777	74.881	75.364	75.126	76.256	77.804	70.848	82.690	77.804	78.486	70.848	74.103
	74.881	80.391	75.126	75.126	74.881	76.041	77.979	70.848	82.486	77.064	78.320	70.848	74.103
	75.364	80.391	75.126	75.596	74.370	75.821	77.904	69.575	82.568	77.979	78.151	71.604	74.103
8	74.881	80.391	74.881	75.126	74.630	76.041	77.804	69.575	82.276	77.442	78.151	70.444	73.248
	74.370	71.604	73.827	72.627	71.604	73.248	75.821	79.987	70.444	71.604	76.465	79.417	69.105
	71.235	70.848	73.248	73.248	71.959	73.248	76.669	80.902	71.235	71.959	75.821	78.648	71.235
	72.627	71.604	74.103	73.248	71.235	75.126	75.126	79.707	71.604	71.959	74.370	79.117	68.609
9	72.299	72.627	73.542	72.627	72.943	75.364	77.255	79.848	72.627	71.604	76.669	78.964	69.575
	60.902	59.563	63.085	64.828	59.563	59.563	64.000	70.021	59.563	57.979	60.902	66.923	57.979
	62.062	60.902	64.000	64.828	59.563	60.902	64.000	70.021	57.979	59.563	59.563	66.923	56.041
	59.563	59.563	62.062	64.828	57.979	59.563	63.085	69.105	53.542	56.041	57.979	65.584	50.021
10	73.542	72.299	76.669	73.827	71.235	66.923	65.584	71.604	56.041	57.979	59.563	65.584	53.542
	73.827	72.943	76.041	73.827	71.959	75.126	77.255	81.266	71.959	73.248	77.804	79.563	71.235
	71.959	74.370	74.630	74.103	71.959	74.630	77.625	80.902	70.444	72.627	75.821	78.807	70.021
	71.235	73.542	74.881	72.627	71.959	72.943	77.255	81.730	71.235	72.627	75.821	80.124	69.575
11	72.299	73.542	74.881	73.542	72.527	75.821	76.869	81.385	72.299	74.103	77.442	78.964	70.444
	77.442	81.266	79.417	78.320	77.442	78.648	80.777	80.902	82.890	78.320	81.147	80.777	75.364
	77.442	81.266	78.320	77.979	77.255	79.563	80.777	80.902	82.790	78.486	80.777	81.385	75.364
	77.625	81.025	78.320	77.804	77.979	78.964	80.902	81.385	83.085	78.807	81.025	81.501	75.596
12	77.442	81.025	77.979	77.979	77.064	78.320	81.730	81.501	82.690	78.807	81.025	82.890	75.821
	80.124	81.501	78.648	78.320	81.616	81.730	79.417	77.442	85.138	81.730	80.391	78.151	78.151
	78.320	82.988	78.151	79.269	84.341	81.147	80.124	75.821	84.588	81.385	80.391	77.625	77.804
	80.391	83.370	77.255	79.563	85.289	81.730	80.391	77.625	85.138	80.902	80.321	75.596	77.625
13	78.807	82.062	77.804	77.064	80.902	81.025	79.707	77.804	84.000	82.170	80.391	77.255	77.625
	67.522	66.923	66.923	65.584	70.021	71.235	67.522	70.848	71.959	76.465	70.444	72.627	71.235
	67.522	66.923	69.575	69.105	67.522	70.848	66.923	70.444	70.021	76.869	71.235	69.105	70.021
	68.082	69.105	68.082	66.523	70.848	70.848	67.522	70.848	70.848	76.256	69.105	71.235	70.444

Figure 19. Data readout

MICROPHONE POSITIONS

	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
85	63.085	62.062	64.000	63.085	64.000	62.062	68.082	64.000	63.085	59.563	68.609	60.902	63.085	64.000	65.584
85	62.062	62.062	63.085	62.062	62.062	62.062	68.082	65.584	64.828	63.085	70.021	64.828	63.085	63.085	66.279
85	63.085	64.000	64.000	62.062	64.828	62.062	68.609	66.279	64.828	64.828	70.848	66.279	63.085	64.000	67.522
28	65.584	63.085	64.828	62.062	67.522	64.828	68.609	68.609	66.279	64.000	70.021	65.584	62.062	62.062	66.279
25	80.258	73.542	71.959	77.064	76.041	75.126	72.627	75.126	74.630	72.943	71.959	70.021	71.604	70.021	69.105
58	80.258	73.542	72.627	77.255	76.256	74.103	71.604	74.103	72.943	71.959	72.299	70.444	71.604	72.299	70.021
50	79.707	72.627	71.559	77.064	75.126	74.370	71.959	74.881	73.248	71.235	70.848	69.105	70.848	70.021	69.575
48	79.848	73.827	71.235	77.064	75.821	74.103	70.021	73.827	73.542	70.848	71.604	70.021	70.848	70.848	69.105
63	60.902	64.828	70.444	62.062	62.062	65.584	67.522	57.979	59.563	57.979	60.902	53.542	56.041	57.979	59.563
02	62.062	64.828	71.235	60.902	59.563	62.062	66.279	56.041	56.041	59.563	60.902	53.542	57.979	59.563	60.902
62	62.062	65.584	73.827	57.979	57.979	63.085	65.584	53.542	57.979	59.563	64.828	57.979	60.902	59.563	62.062
02	63.085	66.279	69.105	60.902	59.563	67.522	65.584	57.979	57.979	60.902	63.085	56.041	59.563	62.062	59.563
00	64.828	64.000	62.062	70.021	68.082	64.828	64.000	75.596	70.444	69.105	68.082	73.248	69.105	66.279	64.000
52	65.584	64.828	64.000	71.235	68.609	65.584	64.000	75.821	68.609	68.082	65.584	73.248	70.021	67.522	65.584
28	66.923	64.828	64.000	70.444	68.082	64.828	64.000	76.041	70.021	69.105	66.923	72.943	69.575	66.923	63.085
00	64.828	64.000	64.828	70.444	69.575	65.584	64.000	75.126	70.021	67.522	65.584	73.248	69.575	67.522	65.584
55	77.442	75.596	75.364	75.126	75.821	73.542	74.370	74.370	73.542	73.248	73.827	72.943	73.542	72.943	72.627
56	77.442	75.364	75.126	75.364	74.630	73.827	74.103	75.364	74.370	73.248	73.542	72.299	73.248	71.959	70.848
55	77.064	74.881	75.596	75.596	75.364	74.103	73.542	73.827	73.542	71.959	73.248	71.959	72.299	72.299	72.299
41	77.804	75.821	75.126	75.596	75.821	73.542	74.103	74.370	73.542	72.299	73.542	72.299	72.943	72.299	71.959
59	74.103	71.604	71.604	81.025	73.248	72.627	72.943	78.648	74.630	70.848	71.235	72.627	75.126	74.103	70.848
48	73.542	72.943	71.604	80.391	73.827	72.627	74.630	78.807	74.630	71.959	71.959	74.881	75.126	73.248	70.444
43	73.827	71.604	70.021	80.650	71.959	72.627	73.248	77.442	74.630	71.959	72.627	74.630	74.630	74.103	70.848
35	74.370	70.848	71.559	79.987	71.604	72.299	73.542	78.964	74.630	70.444	73.248	73.248	75.364	74.630	71.235
21	76.041	72.299	72.299	78.807	78.486	73.827	71.604	76.869	76.465	71.604	74.103	71.604	74.103	70.444	73.542
56	76.256	72.299	70.848	78.907	78.648	72.943	70.444	76.869	76.465	71.235	74.103	71.604	73.827	71.235	74.103
59	76.256	72.299	71.959	78.320	78.320	72.943	70.021	76.669	76.041	69.575	73.248	71.235	73.542	71.235	74.103
59	76.041	72.299	71.959	79.117	78.486	73.248	69.575	76.465	76.041	70.444	73.248	71.235	73.542	71.604	73.827
21	80.777	74.881	75.364	75.126	76.256	77.804	70.848	82.690	77.804	78.486	70.848	74.103	75.596	72.299	75.364
31	80.391	75.126	75.126	74.881	76.041	77.979	70.848	82.486	77.064	78.320	70.848	74.103	75.821	72.943	76.041
34	80.391	75.126	75.596	74.370	75.821	77.904	69.575	82.568	77.979	78.151	71.604	74.103	75.364	71.959	75.596
31	80.391	74.881	75.126	74.630	76.041	77.804	69.575	82.276	77.442	78.151	70.444	73.248	75.596	72.299	75.596
0	71.604	73.827	72.627	71.604	73.248	75.821	79.987	70.444	71.604	76.465	79.417	69.105	72.299	73.248	74.103
15	70.848	73.248	73.248	71.959	73.248	76.669	80.902	71.235	71.959	75.821	78.648	71.235	73.542	75.126	76.465
17	71.604	74.103	73.248	71.235	75.126	75.126	79.707	71.604	71.959	74.370	79.117	68.609	72.943	75.126	74.103
19	72.627	73.542	72.627	72.943	75.364	77.255	79.848	72.627	71.604	76.669	78.964	69.575	74.881	73.827	75.364
12	59.563	63.085	64.828	59.563	59.563	64.000	70.021	59.563	57.979	60.902	66.923	57.979	60.902	62.062	63.085
2	60.902	64.000	64.828	59.563	60.902	64.000	70.021	57.979	59.563	59.563	66.923	56.041	59.563	63.085	63.085
3	59.563	62.062	64.828	57.979	59.563	63.085	69.105	53.542	56.041	57.979	65.584	50.021	56.041	59.563	74.881
2	72.299	76.669	73.827	71.235	66.923	65.584	71.604	56.041	57.979	59.563	65.584	53.542	56.041	60.902	63.085
7	72.943	76.041	73.827	71.959	75.126	77.255	81.266	71.959	73.248	77.804	79.563	71.235	74.881	74.881	76.256
9	74.370	74.630	74.103	71.959	74.630	77.625	80.902	70.444	72.627	75.821	78.807	70.021	74.103	75.126	77.255
5	73.542	74.881	72.627	71.959	72.943	77.255	81.730	71.235	72.627	75.821	80.124	69.575	73.248	76.041	76.465
9	73.542	74.881	73.542	72.627	75.821	75.866	81.385	72.299	74.103	77.442	78.964	70.444	74.881	74.881	74.881
2	81.266	79.417	78.320	77.442	78.648	80.777	80.902	82.890	78.320	81.147	80.777	75.364	78.648	76.869	78.320
2	81.266	78.320	77.579	77.255	79.563	80.777	80.902	82.790	78.486	80.777	81.385	75.364	78.648	76.669	79.707
5	81.025	78.320	77.804	77.979	78.964	80.902	81.385	83.085	78.807	81.025	81.501	75.596	77.979	77.255	78.964
2	81.025	77.979	77.979	77.064	78.320	81.730	81.501	82.690	78.807	81.025	82.890	75.821	78.964	77.625	78.486
4	81.501	78.648	78.320	81.616	81.730	79.417	77.442	85.138	81.730	80.391	78.151	78.151	80.650	77.625	76.486
0	82.988	78.151	79.269	84.341	81.147	80.124	75.821	84.588	81.385	80.391	77.625	77.804	80.521	77.625	78.807
1	83.370	77.255	79.563	85.289	81.730	80.391	77.625	85.138	80.902	80.521	75.596	77.625	79.707	77.255	78.151
7	82.062	77.804	77.064	80.902	81.025	79.707	77.804	84.000	82.170	80.391	77.255	77.625	79.848	76.869	78.486
2	66.923	66.923	65.584	70.021	71.235	67.522	70.848	71.959	76.465	70.444	72.627	71.235	69.105	71.235	71.235
2	66.923	69.575	69.105	67.522	70.848	66.923	70.444	70.021	76.869	71.235	69.105	70.021	70.444	70.444	67.522
2	69.105	68.082	66.523	70.848	70.848	67.522	70.848	70.848	76.256	69.105	71.235	70.444	68.609	70.444	70.444

Data readout

	MICROPHONE POSITIONS												
	1	2	3	4	5	6	7	8	9	10	11	12	13
14	67.522	66.923	69.105	69.575	69.105	71.235	68.609	70.021	71.235	76.465	71.235	69.575	69.575
	77.625	76.669	81.266	74.630	77.804	76.869	71.235	75.364	66.923	70.848	66.923	76.465	70.444
	71.959	73.827	77.379	74.630	74.370	70.444	73.542	75.364	69.105	71.604	69.575	77.804	69.575
	70.021	75.126	78.649	73.542	72.299	70.021	72.299	73.248	69.105	70.021	66.279	72.299	63.085
15	69.105	75.821	78.648	73.248	73.542	77.442	73.542	72.943	69.105	68.082	68.609	74.881	70.444
	72.627	81.025	77.442	72.943	74.370	77.804	73.248	73.248	70.848	66.923	69.105	72.627	68.082
	72.943	80.521	77.255	73.248	74.881	78.648	73.542	71.959	71.235	67.522	69.575	72.627	67.522
	72.627	80.258	77.255	73.542	74.103	78.486	73.248	72.627	70.848	67.522	69.575	72.627	67.522
16	73.248	79.417	77.255	73.542	74.370	77.804	73.542	71.235	70.848	66.923	70.021	72.299	66.923
	78.486	81.842	80.777	79.848	80.391	83.735	81.842	78.807	78.320	80.521	83.825	82.062	78.807
	78.486	81.266	80.650	80.124	80.650	83.370	82.276	78.807	78.807	81.025	83.825	81.730	78.648
	77.804	80.521	80.124	79.563	80.777	83.181	82.382	78.648	78.807	80.777	84.341	82.062	78.486
17	77.442	80.521	80.124	78.964	79.987	83.276	82.062	78.807	78.807	81.147	84.424	81.266	77.804
	81.266	79.987	76.256	76.669	78.320	77.064	76.465	75.364	78.486	76.465	74.103	73.542	75.596
	80.124	80.124	75.126	75.126	78.320	76.669	75.821	75.821	77.979	75.364	73.827	73.542	76.041
	80.258	80.521	76.465	77.625	78.964	76.869	75.821	74.881	77.804	76.256	74.103	73.248	75.821
18	79.848	79.269	74.630	74.630	78.807	76.869	76.669	75.596	77.442	74.881	72.943	71.959	75.126
	80.777	83.181	84.668	81.266	82.486	84.829	84.172	80.902	79.563	81.842	84.828	82.790	78.964
	79.848	83.181	84.506	81.147	82.062	84.906	83.825	80.902	79.417	81.616	84.668	82.588	78.486
	79.707	83.463	84.589	81.395	82.382	95.138	83.913	81.395	79.417	81.730	84.506	82.890	78.151
19	80.391	83.276	84.668	81.266	82.382	84.984	83.735	80.521	79.269	81.730	84.749	83.085	78.648
	81.730	82.890	80.650	80.902	85.062	82.890	82.276	82.170	84.906	82.890	82.170	81.501	79.848
	82.170	83.463	79.707	79.848	82.276	82.790	82.690	82.382	85.656	82.382	82.170	82.968	80.124
	80.521	83.370	79.987	80.391	84.828	82.890	81.842	82.062	85.364	82.988	81.951	82.486	79.269
20	81.953	82.690	79.707	79.587	82.486	82.276	82.988	82.276	84.506	82.890	81.953	83.735	79.269
	86.007	87.107	86.345	84.984	85.584	87.227	86.607	85.138	86.477	86.076	86.984	86.798	82.890
	85.868	87.580	85.727	85.289	87.464	87.167	86.734	84.828	86.542	86.477	86.542	85.798	83.181
	84.984	87.167	86.860	85.214	86.923	87.637	86.734	85.364	86.860	86.007	86.860	86.007	83.645
21	85.727	87.522	86.076	85.438	86.212	87.907	86.734	84.984	87.346	86.007	86.984	86.345	82.890
	86.671	88.558	86.477	85.656	89.105	89.621	87.694	84.749	87.637	86.477	87.107	87.227	86.734
	86.345	88.402	87.045	86.212	89.201	89.621	87.464	85.933	87.807	86.144	87.287	87.227	86.345
	86.411	89.057	85.734	85.214	89.666	89.711	87.046	86.734	87.464	86.734	87.405	87.580	86.542
22	86.477	88.558	86.607	85.511	89.105	89.457	88.190	85.798	87.167	86.984	87.464	88.297	84.984
	86.411	88.811	86.860	85.727	89.889	89.437	87.464	87.167	87.694	86.212	87.287	86.860	86.279
	86.923	86.558	87.167	86.477	89.811	90.021	87.522	86.671	88.297	86.076	87.580	87.227	86.345
	86.860	89.660	87.046	86.212	90.361	89.999	87.405	86.542	87.807	86.212	87.107	87.287	86.007
23	87.167	88.660	86.923	85.668	89.711	89.529	87.637	85.868	88.028	86.279	87.287	87.167	86.542

Figure 19 (Continued)

MICROPHONE POSITIONS

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
522	66.923	69.105	69.575	69.105	71.235	68.609	70.021	71.235	76.465	71.235	69.575	69.575	69.575	70.444	65.584
625	76.665	81.266	74.630	77.804	76.869	71.235	75.364	66.923	70.848	66.923	76.465	70.444	69.105	68.609	71.959
959	73.827	77.779	74.630	74.370	70.444	73.542	75.364	69.105	71.604	69.575	77.804	69.575	68.082	70.021	73.827
021	75.126	72.649	73.542	72.299	70.021	72.299	73.248	69.105	70.021	66.279	72.299	63.085	66.279	71.235	70.021
105	75.821	78.648	73.248	73.542	77.442	73.542	72.943	69.105	68.082	68.609	74.881	70.444	66.923	65.584	67.522
627	81.025	77.442	72.543	74.370	77.804	73.248	73.248	70.848	66.923	69.105	72.627	68.082	65.584	69.105	66.923
943	80.521	77.255	73.248	74.881	78.648	73.542	71.959	71.235	67.522	69.575	72.627	67.522	66.279	68.609	66.923
627	80.258	77.255	73.542	74.103	78.486	73.248	72.627	70.848	67.522	69.575	72.627	67.522	65.584	68.082	66.279
248	79.417	77.255	73.542	74.370	77.804	73.542	71.235	70.848	66.923	70.021	72.299	66.923	66.279	68.609	66.279
486	81.842	80.777	79.848	80.391	83.735	81.842	78.807	78.320	80.521	83.825	82.062	78.807	79.417	82.790	79.707
486	81.266	80.650	80.124	80.650	83.370	82.276	78.807	78.807	81.025	83.825	81.730	78.648	78.964	82.382	79.417
804	80.521	80.124	79.563	80.777	83.181	82.382	78.648	78.807	80.777	84.341	82.062	78.486	79.417	82.062	79.417
442	80.521	80.124	78.964	79.987	83.276	82.062	78.807	78.807	81.147	84.424	81.266	77.804	79.417	81.730	80.124
266	79.987	76.256	76.669	78.320	77.064	76.465	75.364	78.486	76.465	74.103	73.542	75.596	74.630	73.542	72.627
124	80.124	75.125	75.126	78.320	76.669	75.821	75.821	77.979	75.364	73.827	73.542	76.041	74.370	73.542	71.604
258	80.521	76.465	77.625	78.964	76.869	75.821	74.881	77.804	76.256	74.103	73.248	75.821	73.827	73.248	72.299
848	75.269	74.630	74.630	78.807	76.869	76.669	75.596	77.442	74.881	72.943	71.959	75.126	73.827	72.943	71.235
777	83.181	84.668	81.266	82.486	84.828	84.172	80.902	79.563	81.842	84.828	82.790	78.964	80.391	82.276	81.025
848	83.181	84.506	81.147	82.062	84.906	83.825	80.902	79.417	81.616	84.668	82.588	78.486	80.391	81.953	80.777
707	83.463	84.588	81.395	82.382	85.138	83.913	81.385	79.417	81.730	84.506	82.890	78.151	80.124	82.276	81.147
391	83.276	84.668	81.266	82.382	84.984	83.735	80.521	79.269	81.730	84.749	83.025	78.648	80.391	82.486	80.902
730	82.890	80.650	80.902	85.062	82.890	82.276	82.170	84.906	82.890	82.170	81.501	79.848	81.616	80.124	80.521
170	83.463	79.707	79.848	82.276	82.790	82.690	82.382	85.656	82.382	82.170	82.968	80.124	81.147	80.391	79.707
521	83.370	79.987	80.391	84.828	82.890	81.842	82.062	85.364	82.988	81.953	82.486	79.269	81.501	80.391	79.117
953	82.690	79.707	79.587	82.486	82.276	82.988	82.276	84.506	82.890	81.953	83.735	79.269	80.521	80.650	80.650
007	87.107	86.345	84.984	85.584	87.227	86.607	85.138	86.477	86.076	86.984	86.798	82.890	84.506	85.138	83.825
868	87.580	85.727	85.289	87.464	87.167	86.734	84.828	86.542	86.477	86.542	85.798	83.181	85.062	85.138	84.341
994	87.167	86.860	85.214	86.923	87.637	86.734	85.364	86.860	86.007	86.860	86.007	83.645	84.341	85.214	84.984
727	87.522	86.076	85.438	86.212	87.937	86.734	84.984	87.346	86.007	86.984	86.345	82.890	84.588	85.062	84.424
571	88.558	86.477	85.656	89.105	89.621	87.694	84.749	87.637	86.477	87.107	87.227	86.734	86.076	86.007	84.424
345	88.402	87.045	86.212	89.201	89.621	87.464	85.933	87.807	86.144	87.287	87.227	86.345	86.212	86.345	84.424
411	89.057	86.734	85.214	89.666	89.711	87.046	86.734	87.464	86.734	87.405	87.580	86.542	86.860	85.938	84.906
477	88.558	86.607	85.511	89.105	89.437	88.190	85.798	87.167	86.984	87.464	88.257	84.984	86.671	86.144	84.749
411	88.811	86.860	85.727	89.889	89.437	87.464	87.167	87.694	86.212	87.287	86.860	86.279	86.345	86.144	84.506
923	88.558	87.167	86.477	89.811	90.021	87.522	86.671	88.297	86.076	87.580	87.227	86.345	86.212	86.007	85.214
860	88.660	87.046	86.212	90.361	89.889	87.405	86.542	87.807	86.212	87.107	87.287	86.007	86.798	85.938	84.749
167	88.660	86.923	85.668	89.711	89.529	87.637	85.868	88.028	86.279	87.287	87.167	86.542	86.542	86.212	85.214

(Continued)

treatments have been recorded for the duration of four matrix cycles or 64 readings. The total of 2472 (23 x 64) readings have been printed in row-column format with all readings for each microphone identified by columns. The 92 rows represent the four matrix cycles for each of 23 treatment combinations. This data was now in a form for analysis.

Analysis of Data

Although a visual analysis of the data in Figure 19 revealed certain variations in sound levels observed at different microphone positions, a more comprehensive method of displaying the variations produced by each treatment was through the use of computer developed sound contours based on the average microphone readings (Figure 20) for the four matrix cycles per treatment. These contours will be discussed in the following chapter.

		MICROPHONE POSITIONS												
		1	2	3	4	5	6	7	8	9	10	11	12	13
TREATMENTS	1	63.521	63.454	62.802	63.978	62.318	64.603	62.753	68.346	66.118	64.755	62.869	69.875	64.398
	2	80.445	80.019	73.385	71.345	77.112	75.911	74.426	71.553	74.484	73.591	71.746	71.678	69.898
	3	60.857	62.022	65.380	71.153	60.461	59.792	64.563	66.242	56.386	57.891	59.502	62.429	55.276
	4	63.722	65.540	64.414	63.722	70.536	69.587	65.206	64.000	75.646	69.774	68.454	66.543	73.172
	5	76.504	77.438	75.416	75.303	75.420	75.409	73.754	74.030	74.483	73.749	72.699	73.540	72.375
	6	72.346	73.961	71.750	71.297	80.513	72.660	72.545	73.591	78.465	74.630	71.303	72.267	73.847
	7	76.404	76.148	72.299	71.766	72.763	78.485	73.240	70.411	76.718	76.253	70.715	73.675	71.419
	8	75.237	80.427	75.004	75.303	74.752	76.040	77.848	70.212	82.510	77.572	78.277	70.936	73.889
	9	72.633	71.671	73.680	72.338	71.935	74.246	76.218	80.111	71.478	71.781	75.831	79.037	69.631
	10	64.017	63.082	66.454	67.078	62.085	61.738	64.167	70.193	56.782	57.891	59.502	66.253	54.396
	11	72.330	73.600	75.108	73.525	72.126	74.630	77.251	81.321	71.484	73.151	76.722	79.364	70.319
	12	77.488	81.146	78.509	78.021	77.435	78.874	81.046	81.172	82.864	78.605	80.993	81.638	75.536
	13	79.410	82.480	77.965	78.554	83.037	81.408	79.910	77.173	84.716	81.546	80.424	77.157	77.801
	14	67.662	67.468	68.421	67.797	69.374	71.041	67.644	70.540	71.016	76.514	70.505	70.636	70.319
	15	72.177	75.361	79.135	74.012	74.504	73.694	72.655	74.230	68.560	70.139	67.846	75.362	68.387
	16	72.861	80.305	77.302	73.319	74.431	79.185	73.395	72.267	70.945	67.222	69.569	72.545	67.512
	17	73.054	81.038	80.419	79.625	80.451	83.390	82.140	78.767	78.685	80.868	84.103	81.780	78.436
	18	80.374	79.975	75.619	76.012	78.603	76.868	76.194	75.416	77.928	75.742	73.744	73.073	75.646
	19	80.181	83.275	84.608	81.266	82.328	84.964	83.911	80.927	79.416	81.729	84.688	82.838	78.562
	20	81.593	83.103	80.012	80.282	83.663	82.711	82.449	82.222	85.108	82.797	82.061	82.677	79.627
	21	85.647	87.344	86.252	85.231	86.545	87.460	86.702	85.078	86.806	86.142	86.843	86.237	83.151
	22	86.476	88.644	86.716	85.648	89.270	87.597	87.598	85.805	87.519	86.585	87.316	87.583	86.152
	23	86.840	88.672	86.995	86.071	89.693	89.719	87.507	85.562	97.956	86.195	87.315	87.135	86.293

Figure 20. Matrix cycle averages

MICROPHONE POSITIONS

2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
3.454	62.802	63.978	62.318	64.603	62.753	68.346	66.118	64.755	62.869	69.875	64.398	62.829	63.287	66.416
10.018	73.385	71.345	77.112	75.811	74.426	71.553	74.484	73.591	71.746	71.678	69.898	71.226	70.797	69.452
2.029	65.380	71.153	60.461	59.792	64.563	66.242	56.386	57.891	59.502	62.429	55.276	58.621	59.792	60.522
5.540	64.414	63.722	70.536	68.587	65.206	64.000	75.646	65.774	68.454	66.543	73.172	69.569	67.061	64.563
7.438	75.416	75.303	75.420	75.409	73.754	74.030	74.483	73.749	72.699	73.540	72.375	73.008	72.375	71.933
3.961	71.750	71.297	80.513	72.660	72.545	73.591	78.465	74.630	71.303	72.267	73.847	75.061	74.021	70.844
6.148	72.299	71.766	72.763	78.465	73.240	70.411	76.718	76.253	70.715	73.675	71.419	73.754	71.129	73.894
10.427	75.004	75.303	74.752	76.040	77.848	70.212	82.510	77.572	78.277	70.936	73.889	75.594	72.375	75.649
1.671	73.680	72.338	71.935	74.246	76.218	80.111	71.478	71.761	75.831	79.037	69.631	73.417	74.332	75.009
3.082	66.454	67.078	62.085	61.738	64.167	70.198	56.782	57.591	59.502	66.253	54.396	58.137	61.403	66.034
3.600	75.108	73.525	72.126	74.630	77.251	81.321	71.484	73.151	76.722	79.364	70.319	74.278	75.232	76.214
1.146	78.509	78.021	77.435	78.874	81.046	81.172	82.864	78.605	80.993	81.638	75.536	78.560	77.105	78.869
2.480	77.965	78.554	83.037	81.406	79.910	77.173	84.716	81.546	80.424	77.157	77.801	80.181	77.343	78.482
7.468	68.421	67.797	69.374	71.041	67.644	70.540	71.016	76.514	70.505	70.636	70.319	69.433	70.642	68.696
5.361	79.135	74.012	74.504	73.694	72.655	74.230	68.560	70.139	67.846	75.362	68.387	67.597	68.862	70.832
10.305	77.302	73.319	74.431	78.185	73.395	72.267	70.945	67.222	69.569	72.545	67.512	65.931	66.601	66.601
1.038	80.419	79.625	80.451	83.390	82.140	78.767	78.685	80.868	84.103	81.780	78.436	79.304	82.241	79.666
9.975	75.619	76.012	78.603	76.868	76.194	75.416	77.928	75.742	73.744	73.073	75.646	74.164	73.319	71.941
3.275	84.608	81.266	82.328	84.964	83.911	80.927	79.416	81.729	84.688	82.838	78.562	80.324	82.248	80.963
3.103	80.012	80.282	83.663	82.711	82.449	82.222	85.108	82.797	82.061	82.677	79.627	81.196	80.389	79.999
7.344	86.252	85.231	86.545	87.460	86.702	85.078	86.806	86.142	86.843	86.237	83.151	84.624	85.138	84.393
8.644	86.716	85.648	89.270	87.597	87.598	85.805	87.519	86.585	87.316	87.583	86.152	86.455	86.109	84.626
8.672	86.995	86.071	89.693	89.719	87.507	85.562	87.956	86.195	87.315	87.135	86.293	86.474	86.075	84.921

matrix cycle averages

CHAPTER IV. ANALYSIS OF DATA

The information presented in this chapter is in the form of extensive numerical tables and a series of sound contour maps.

Due to the complexity of the format presented in Figure 19, a visual assessment of the data conveyed a minimum of information in terms of the overall changes of the sound environment within the entire laboratory because of the treatments (Table 3). In order to provide a readout format that included the entire laboratory and responded with a space oriented permanent record for comparison, this contour method was developed.

The laboratory floor plan and microphone locations for all treatment contour maps were illustrated in accordance with the layout in Figure 18. The microphone locations are depicted by the bold numerals from one to 16 (see Key, Figure 21). The smaller numerals positioned horizontally in close proximity to each microphone location are the average decibel readings (Figure 20) as monitored by that microphone during a given treatment. The first of this series of 23 treatment contour maps is shown in Figure 22. The contours were developed at one dB intervals from these average levels, and the decibel level of each contour is indicated by a numeral positioned in close parallel relationship to the contour. The machine locations are represented by the light blocks on each treatment contour map. Individual treatments as listed in Table 3 were identified on each map by shading the machine block or blocks used for that particular treatment. For example, Figure 22 shows the contour map for treatment one (Table 3), and it can be noted that the blocks for the north

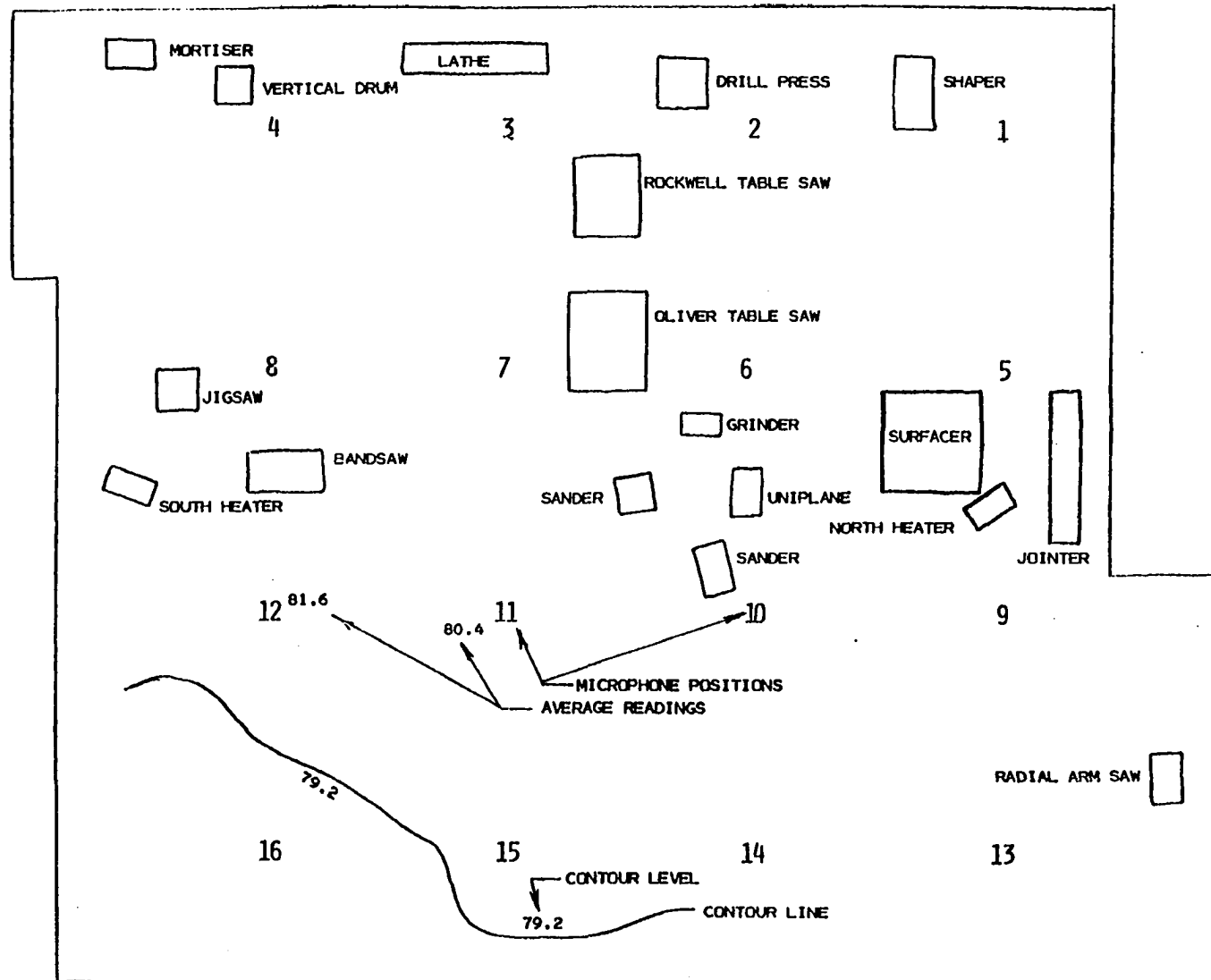


Figure 21. Labeling key for contour map figures

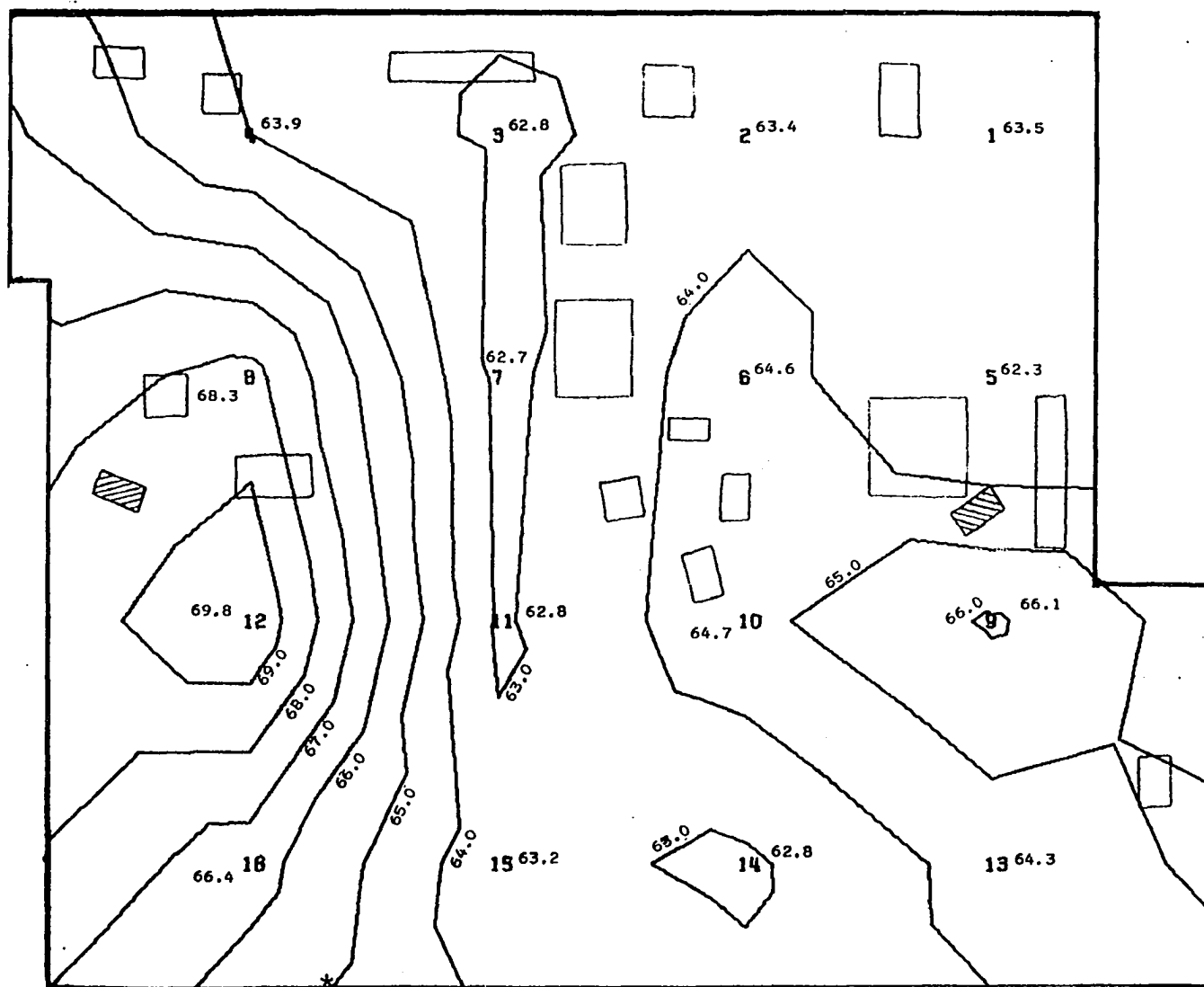


Figure 22. Treatment 1 (* End points on all contour maps represent a reference which was mathematically derived for this particular computer processing.)

and south heaters are shaded, thus showing that they were active. It should be pointed out that the microphones were located five feet above the floor for all treatments, which is generally above the centers of the machines and below the two heaters. The background or ambient noise level measured in the laboratory was 45 dB.

As can be noted from the contour map in Figure 22, the activated heaters raised the noise level 24.8 dB as measured at microphone 12. Although both heaters are identical models, the south heater was the primary source of noise, with the contour lines radiating outward to a point of equal level with the quieter north heater.

The contours produced by treatment two (Figure 23) indicated a predominant source reading of 80.4 dB, which diminished in an even manner to a low of 69.4 dB, as measured in the diagonally opposite corner of the laboratory. Since this laboratory is an enclosure and reverberant rather than free field conditions exist, the even concentric form of the contours radiating away from the source indicated that the sound produced by this shaper did not create pronounced areas of sound addition or subtraction due to internal reflections from the walls of the laboratory. The maximum to minimum sound range for this treatment was 11 dB.

The sound source for treatment three, the mortiser (Figure 24), is located in the southwest corner of the laboratory opposite the shaper. While somewhat lower in level than the shaper, the contours assumed the same even reduction diagonally across the laboratory with a maximum to minimum range of 15.8 dB.

The radial arm saw located in the northeast corner of the

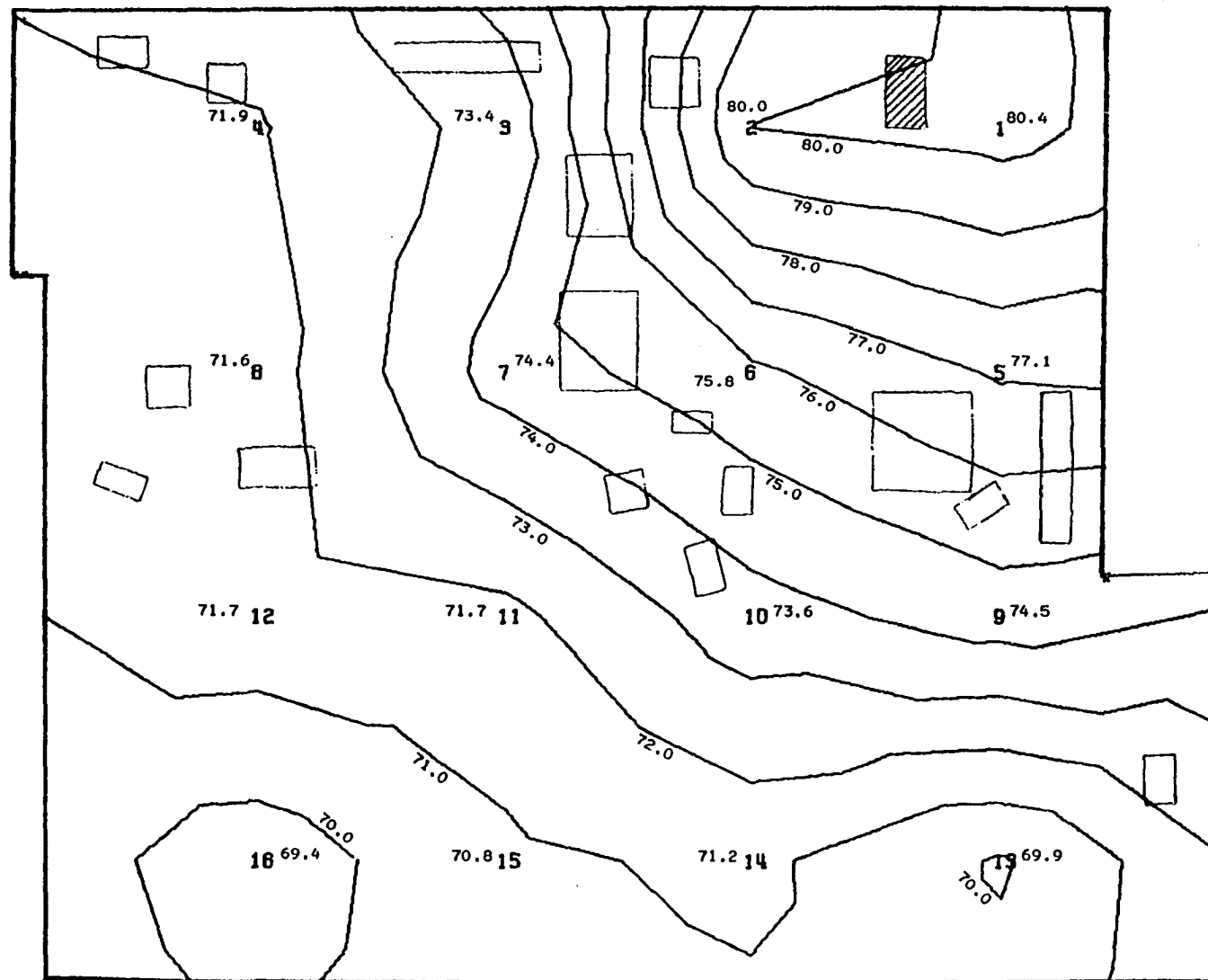


Figure 23. Treatment 2

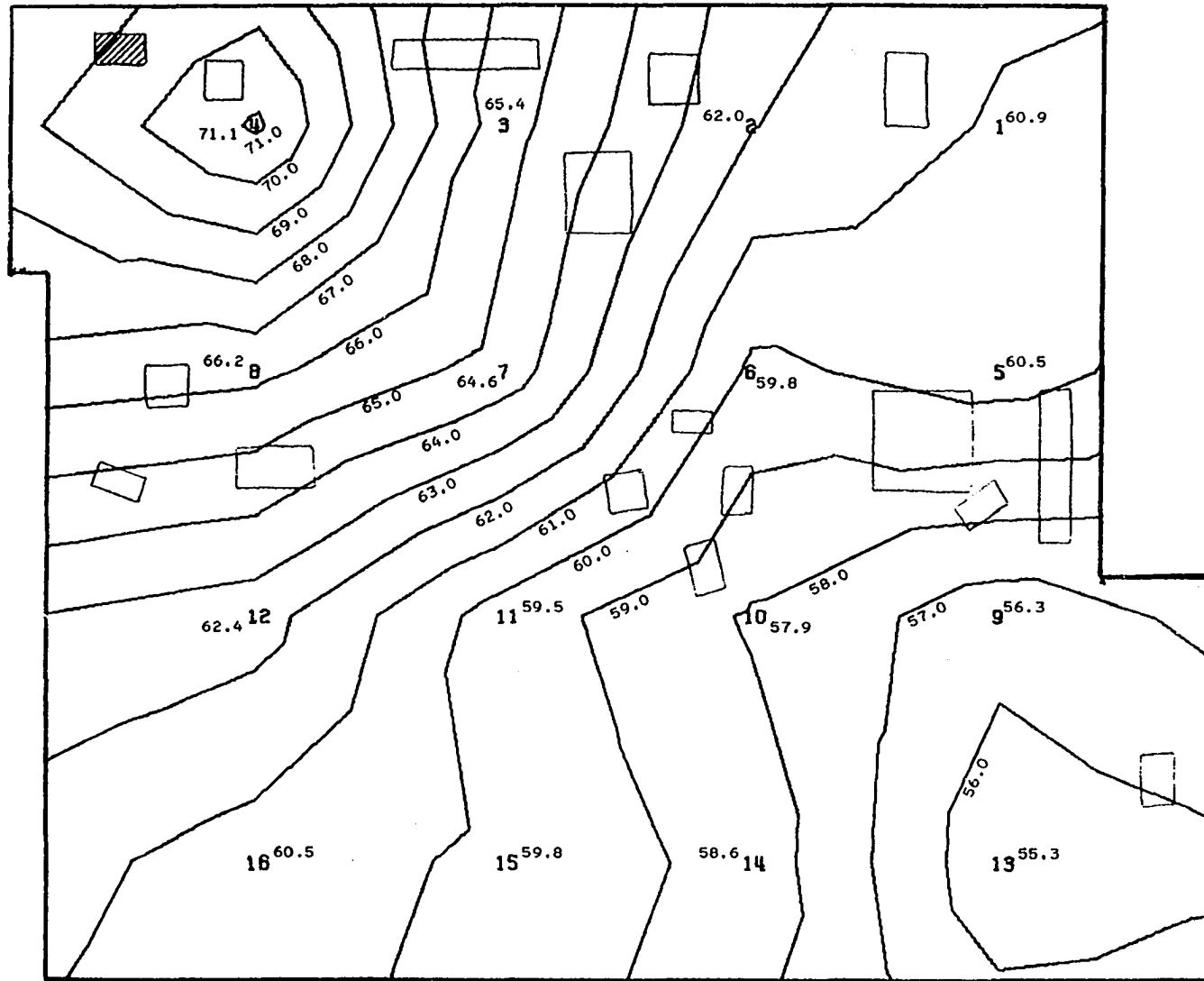


Figure 24. Treatment 3

laboratory (Figure 25), as a single source of noise, again created contours which radiated evenly in a diagonal direction away from the source sound maximum of 75.6 dB to a minimum of 63.7 dB, for a range of 11.9 dB.

The sound source for treatment five (Figure 26) was the dust collector system. This is a multi-locational device with inlets at each machine and floor inlets located on the west and north walls. Since the machines are concentrated toward the west wall, it is logical for the greatest noise to be found along the west wall as shown. Also, the motor and blower units were mounted on the outside of the west wall in the general area between microphones one and two, thus noise and vibration conducted through the exterior wall contributed to the high reading in this area. Since this noise source was noncentralized, a relatively low range of 5.5 dB was recorded.

The jointer in treatment six (Figure 27) is a relatively noisy machine. The contour levels diminished evenly only in the vicinity of the machine. From the center of the laboratory toward the west wall, several areas of contrasting levels developed, thus indicating there were reflections and standing waves. A sound range of 9.6 dB consistent with the previous single source machines was recorded.

The surfacer, a rotating blade machine similar to the jointer, was used for treatment seven (Figure 28). In a manner similar to those of the jointer, the contours radiated in even concentric arcs from the source in the north half of the laboratory. All contours in the south half of the laboratory were close in value, and the range for the entire laboratory was 8.4 dB.

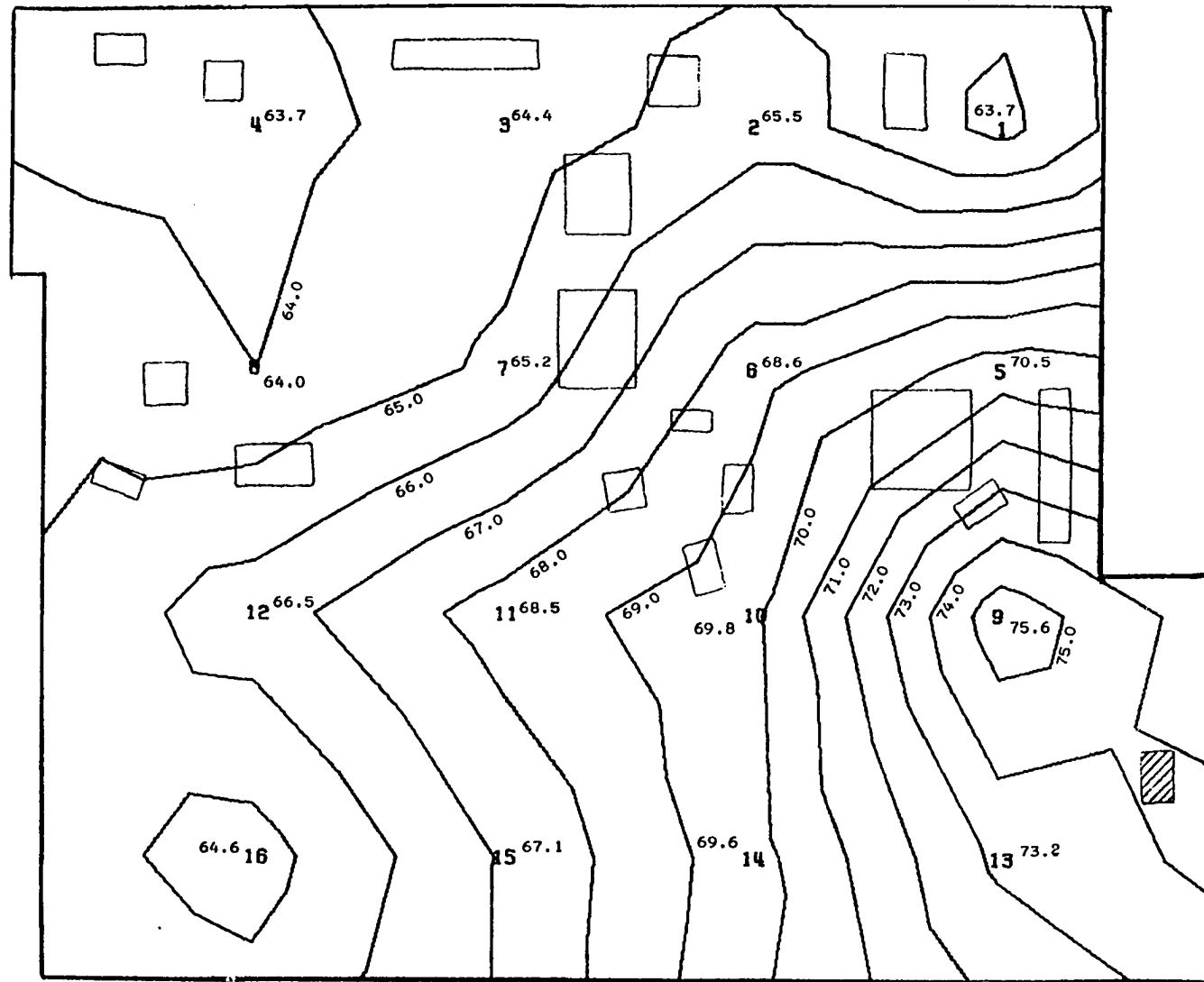


Figure 25. Treatment 4

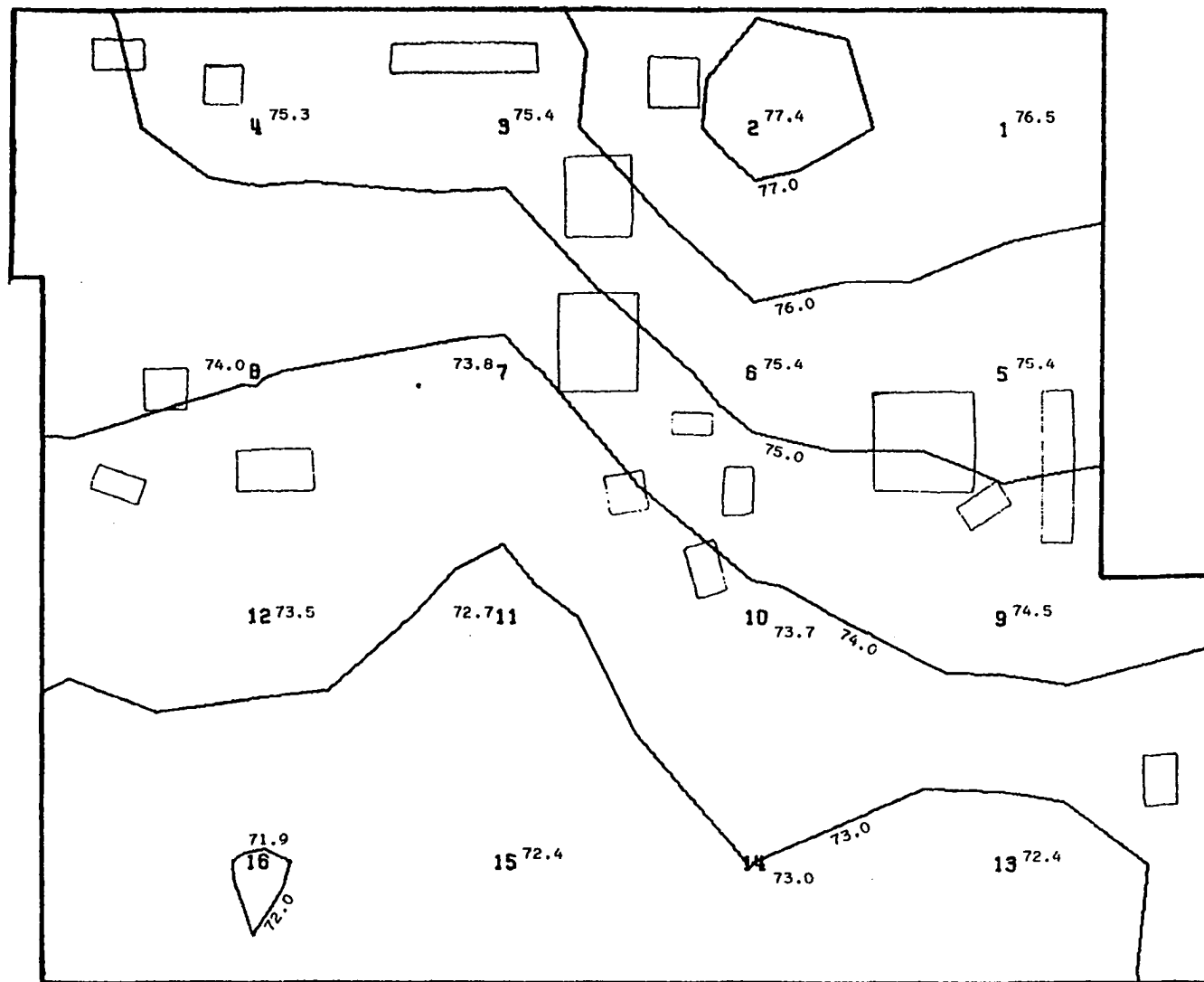


Figure 26. Treatment 5

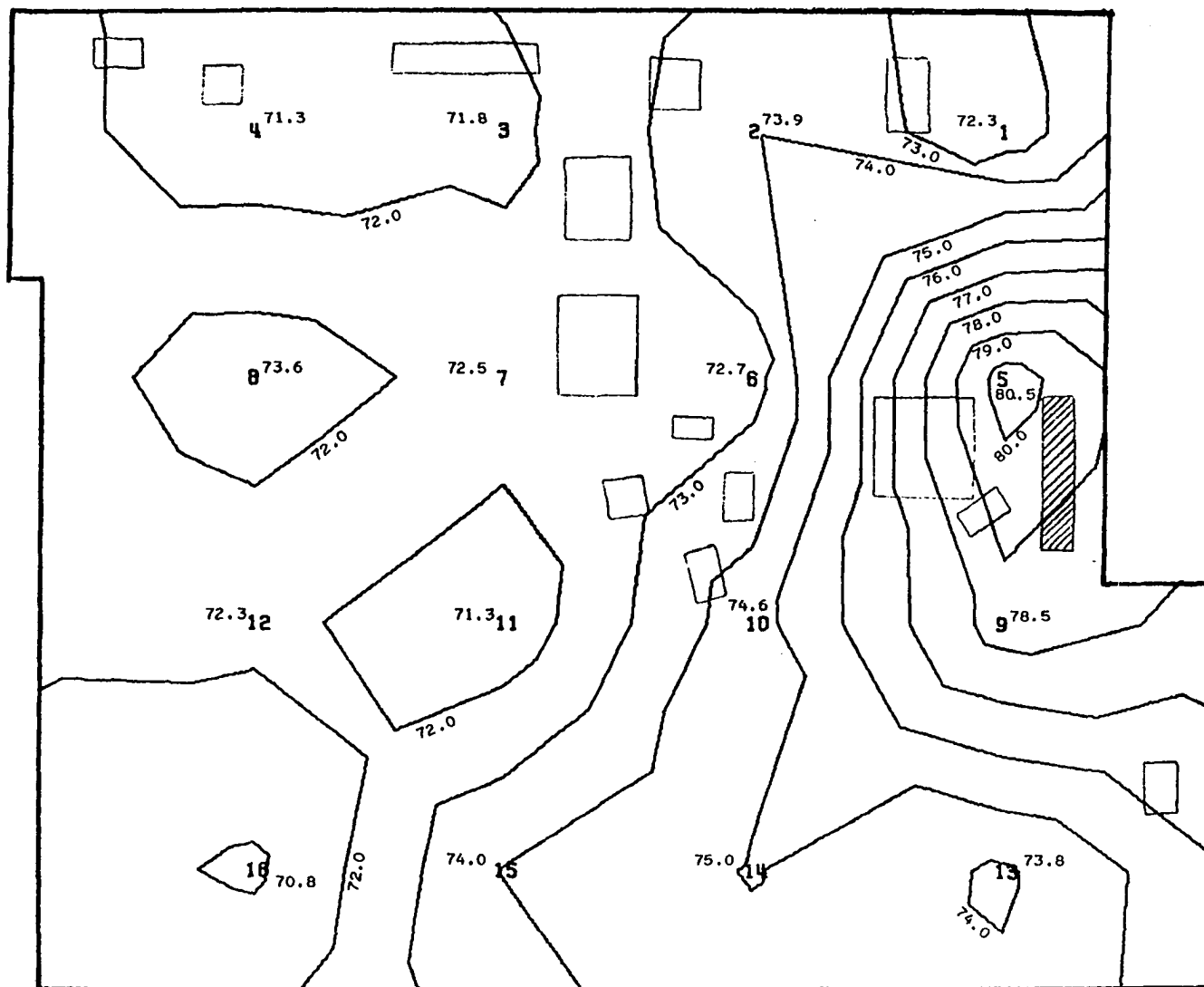


Figure 27. Treatment 6

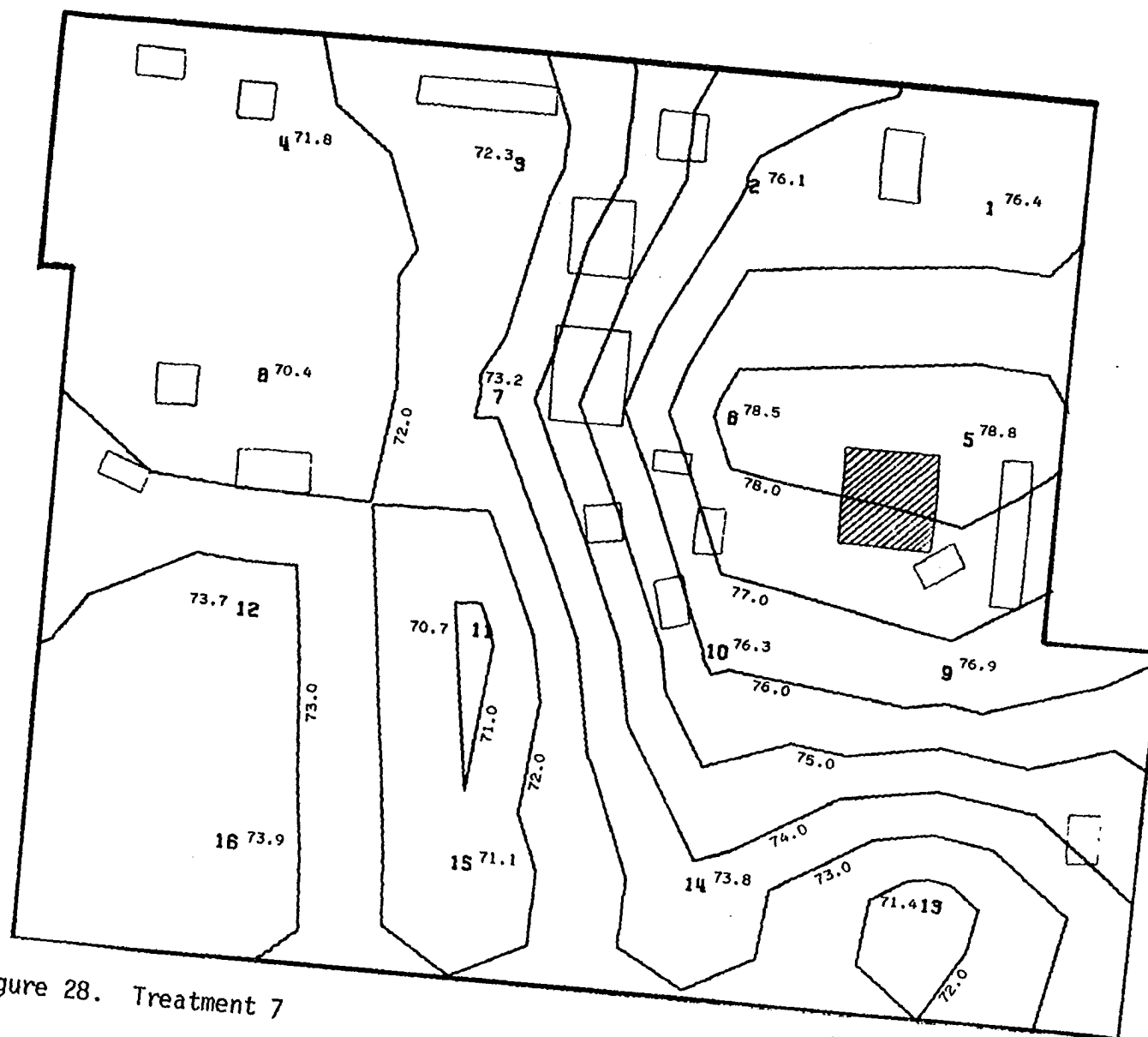


Figure 28. Treatment 7

The contours produced by the uniplane (Figure 29) were unusual. Two distinct high level areas, one around microphone nine and one around microphone two, occupied the north half of the laboratory. These reduced in a complex manner to a point of low reading located in the south center part of the laboratory. Apparently this machine produced a noise which caused diverse standing wave patterns in this enclosure. The maximum to minimum sound level range was 12.3 dB.

The contours shown in Figure 30 were produced by the bandsaw, a relatively loud machine, but with a less penetrating sound than that of the uniplane. The contours were evenly spaced and concentric about the source.

Treatment 10 (Figure 31) used the jigsaw, a relatively quiet machine. The contours show an even reduction in sound level from 70.2 dB to 54.4 dB in the northeast corner of the laboratory. It should be noted that this low level sound source produced the greatest range of change (15.8 dB) yet recorded. This, however, is partially the result of a programming test item that was purposely made to demonstrate the responsiveness of the instrumentation.

Turning to the data readout in Figure 19, it can be seen that the microphone readings for each matrix cycle were consistently equal until the microphone 16 reading of the third matrix cycle for treatment 10. This sudden rise of 10 dB continued through microphone one position of the fourth matrix cycle and gradually returned closer to normal with each successive microphone reading until microphone nine, where the four readings were again equal. This discrepancy was the result of beginning

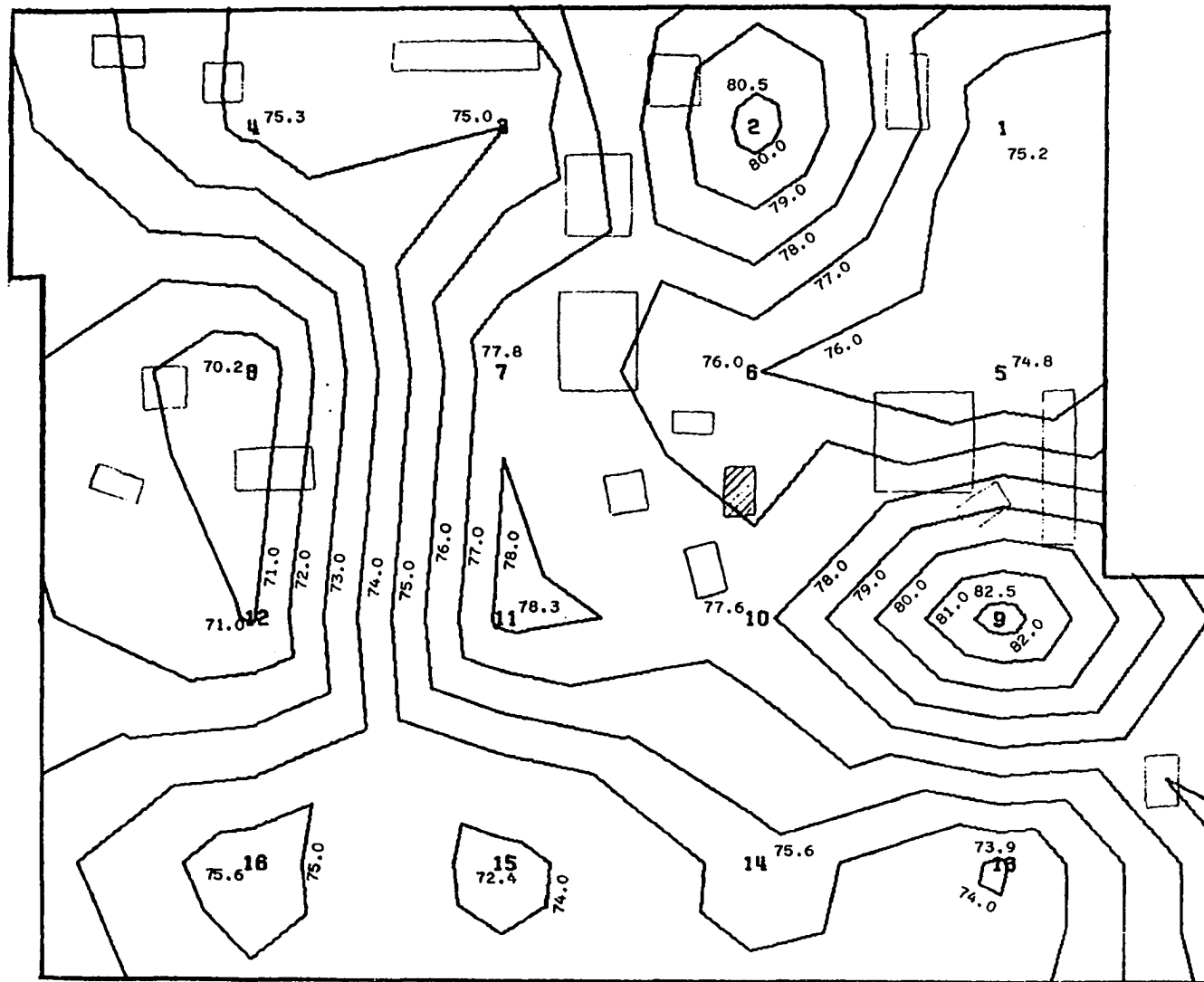


Figure 29. Treatment 8

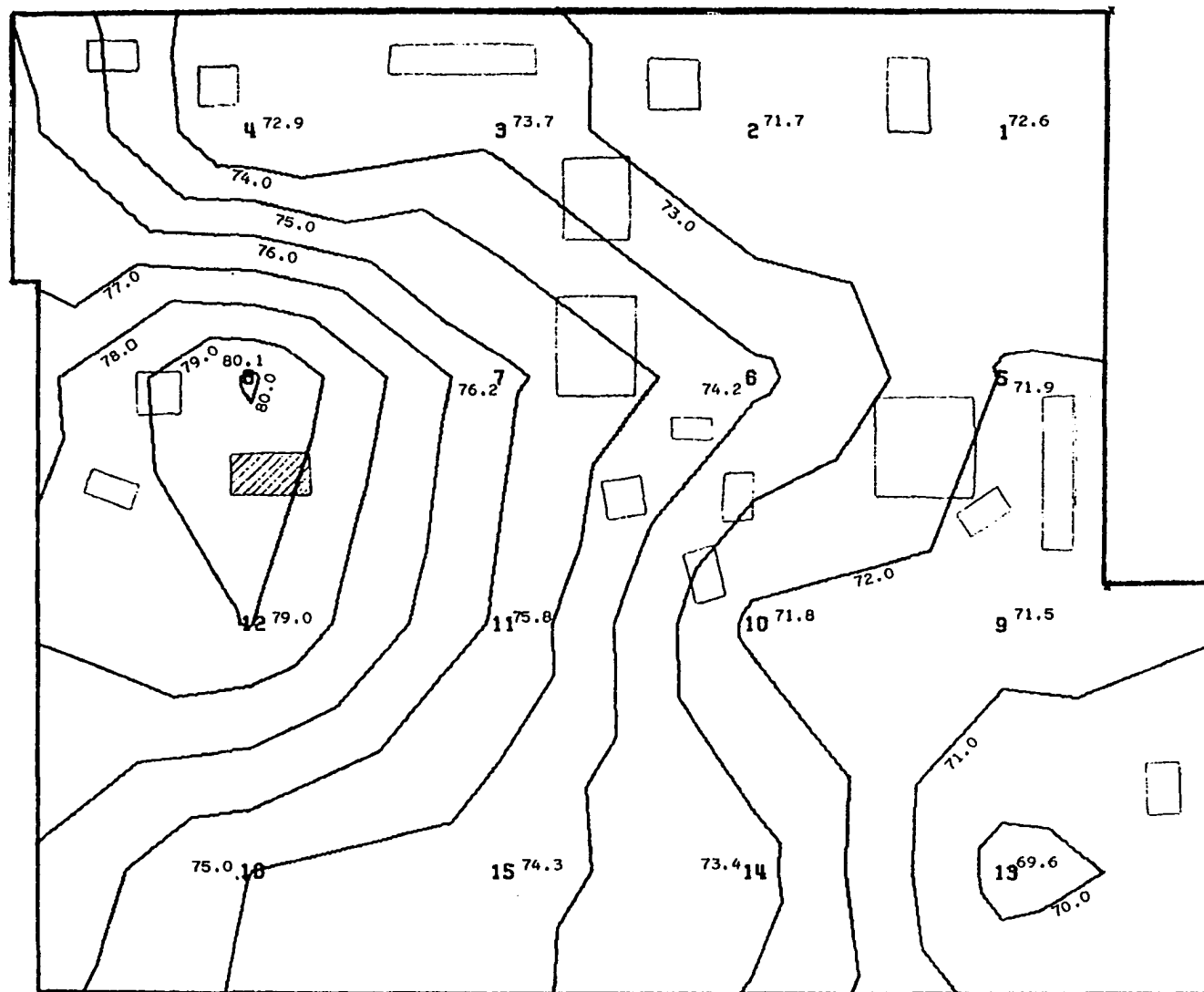


Figure 30. Treatment 9

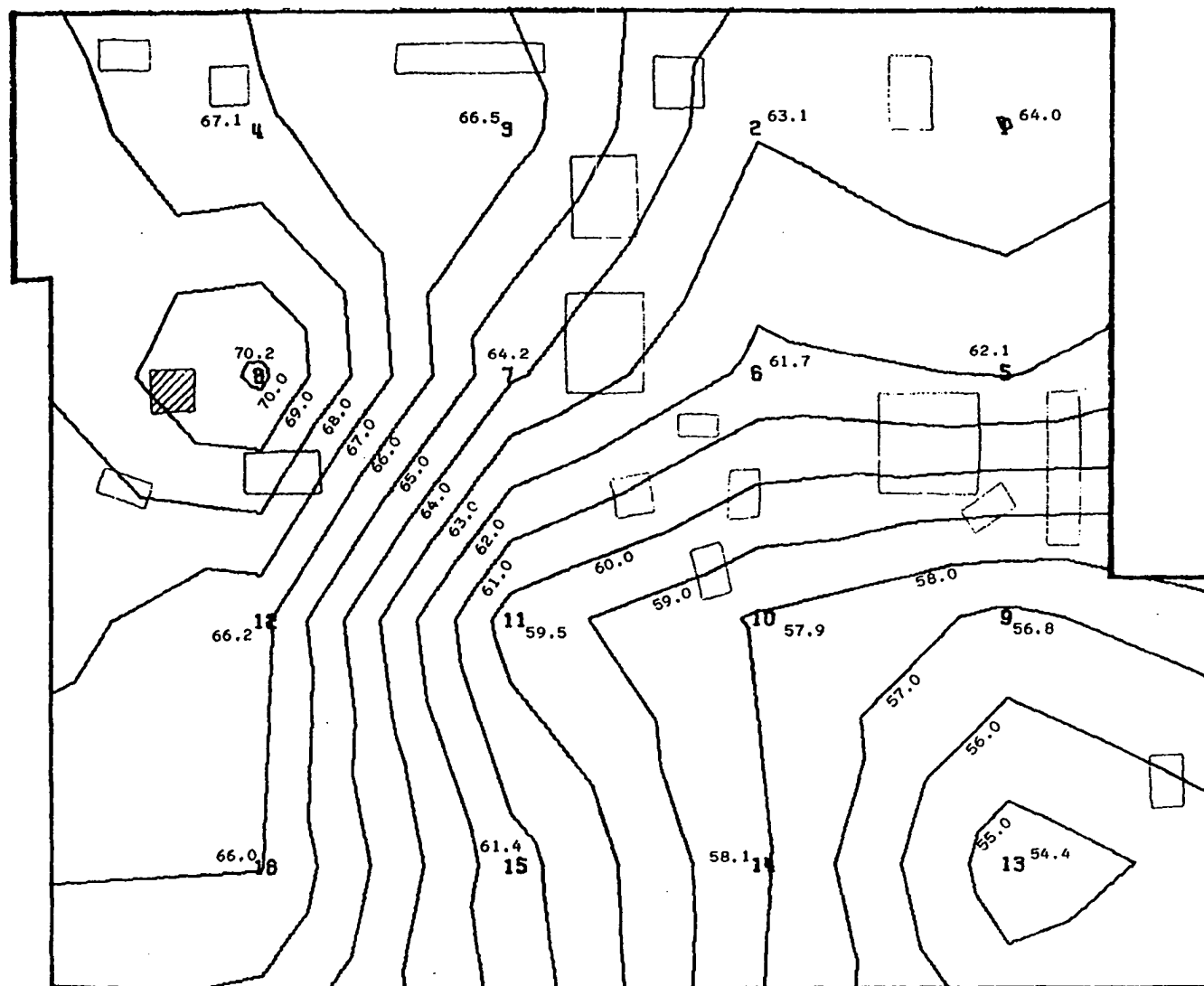


Figure 31. Treatment 10

the next treatment at the end of the third matrix cycle instead of the fourth. This test input feature raised the readings at microphones one and two approximately three dB, but did not affect the readings at microphone eight, where the highest sound level was recorded.

Treatment 11 (Figure 32) is a combination of the two previous treatments. The contour lines radiated from the location of the sources in a manner similar to those in treatment nine, as was expected since the bandsaw was the dominant source. In fact, when comparing each microphone reading for treatments nine and 11, the addition of the jigsaw for treatment 11 caused incremental changes which were within the limits of changes due to random noise in the laboratory. It can be noted in Figure 3 that the addition of two sound levels which were separated by 10 or more dB increased the reading by .4 dB or less, thus substantiating the consistency of the instrumentation system.

The contours of treatment 12 (Figure 33) resulted from the addition of the uniplane to the jigsaw and bandsaw of the previous treatment. When comparing these contours with those of Figure 8, the characteristic pattern around microphone nine was again developed by this combination treatment. Apparently the location and type of sound produced by the uniplane is such that this identifying pattern was produced. Elsewhere in the laboratory the sound levels were substantially raised over those observed in treatment 11. As a result, the maximum-minimum range for treatment 12 (5.8 dB) was substantially reduced from the 11 dB observed for treatment 11.

The combination of the three machines used for treatment 13 (Figure

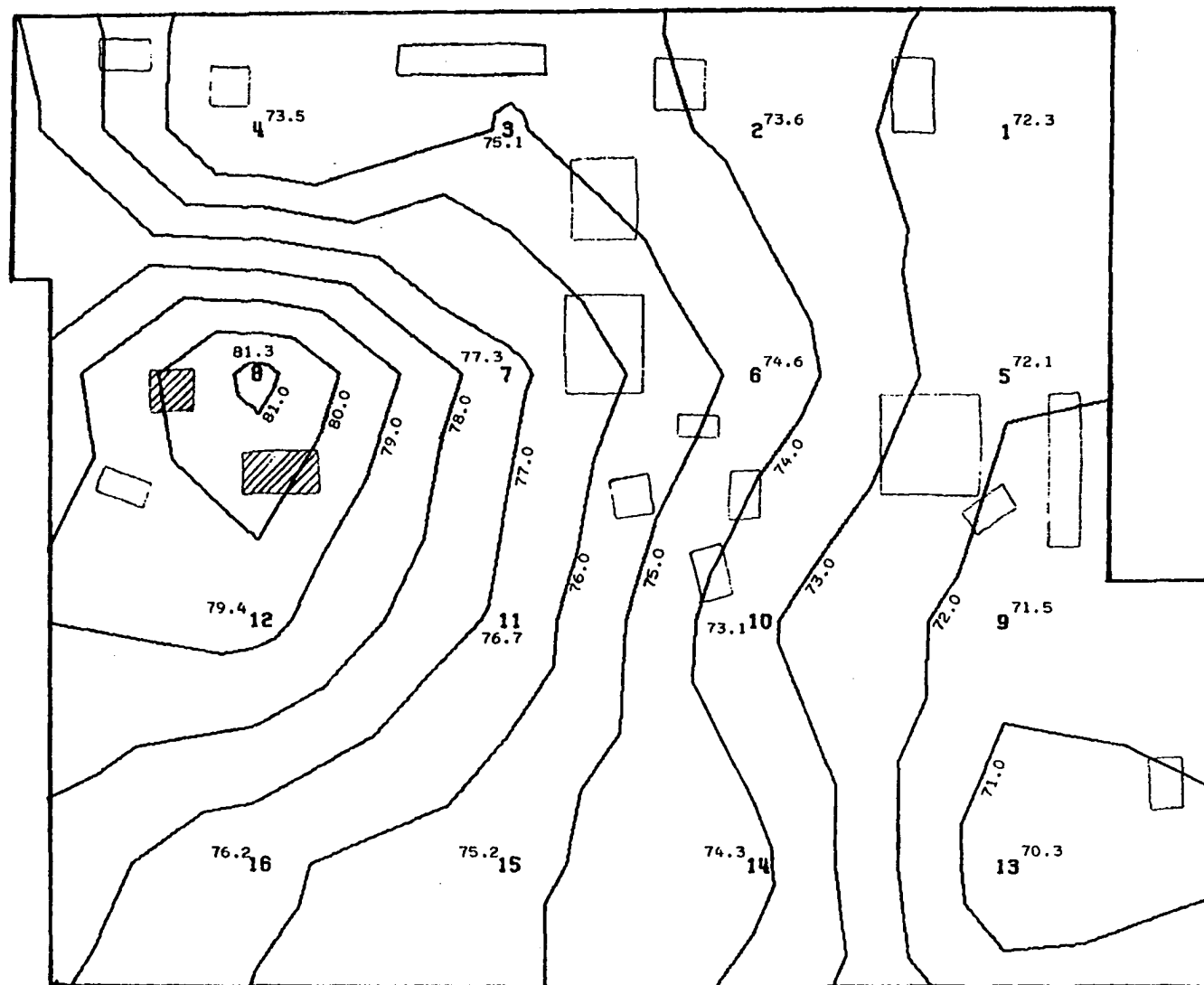


Figure 32. Treatment 11

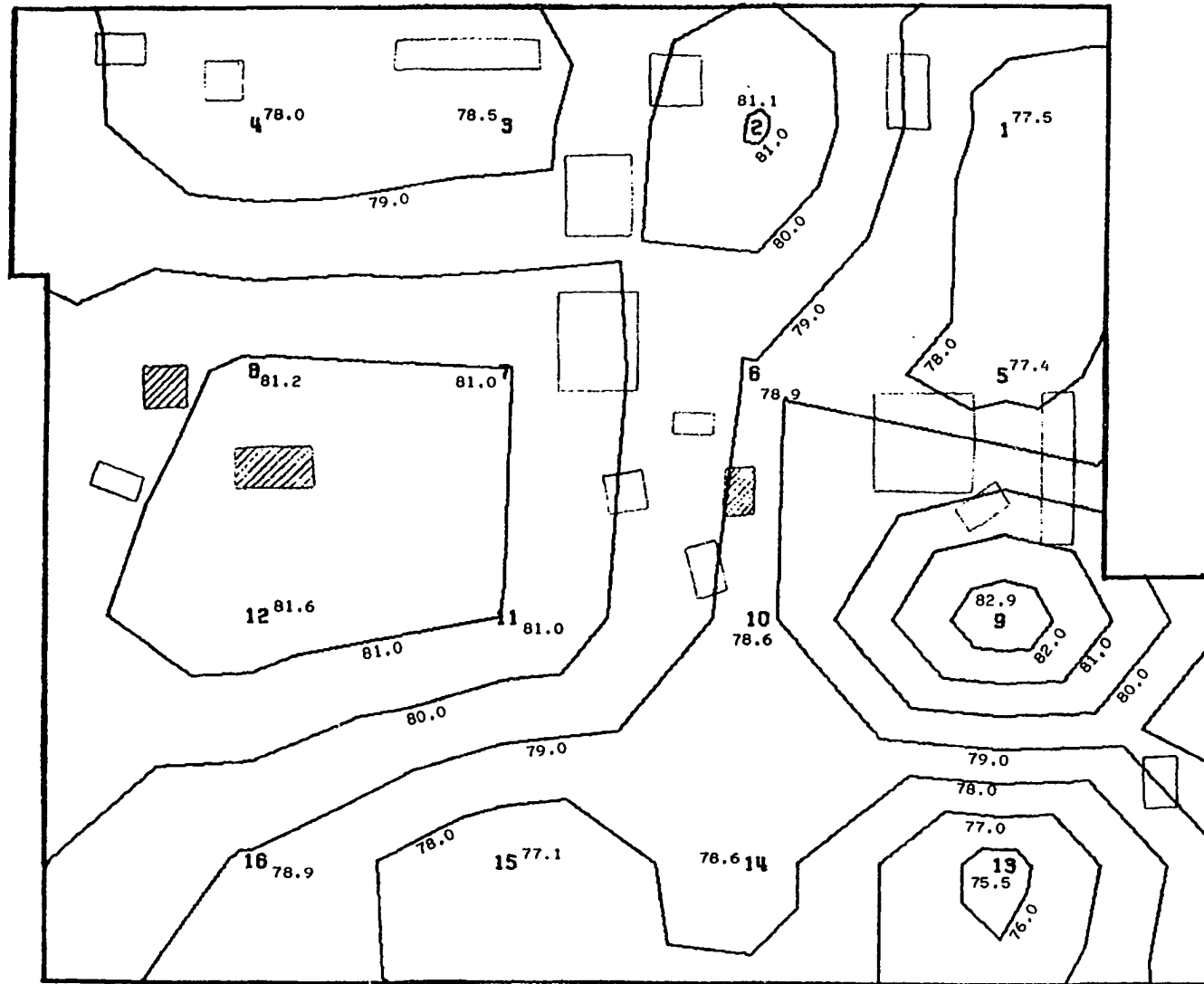


Figure 33. Treatment 12

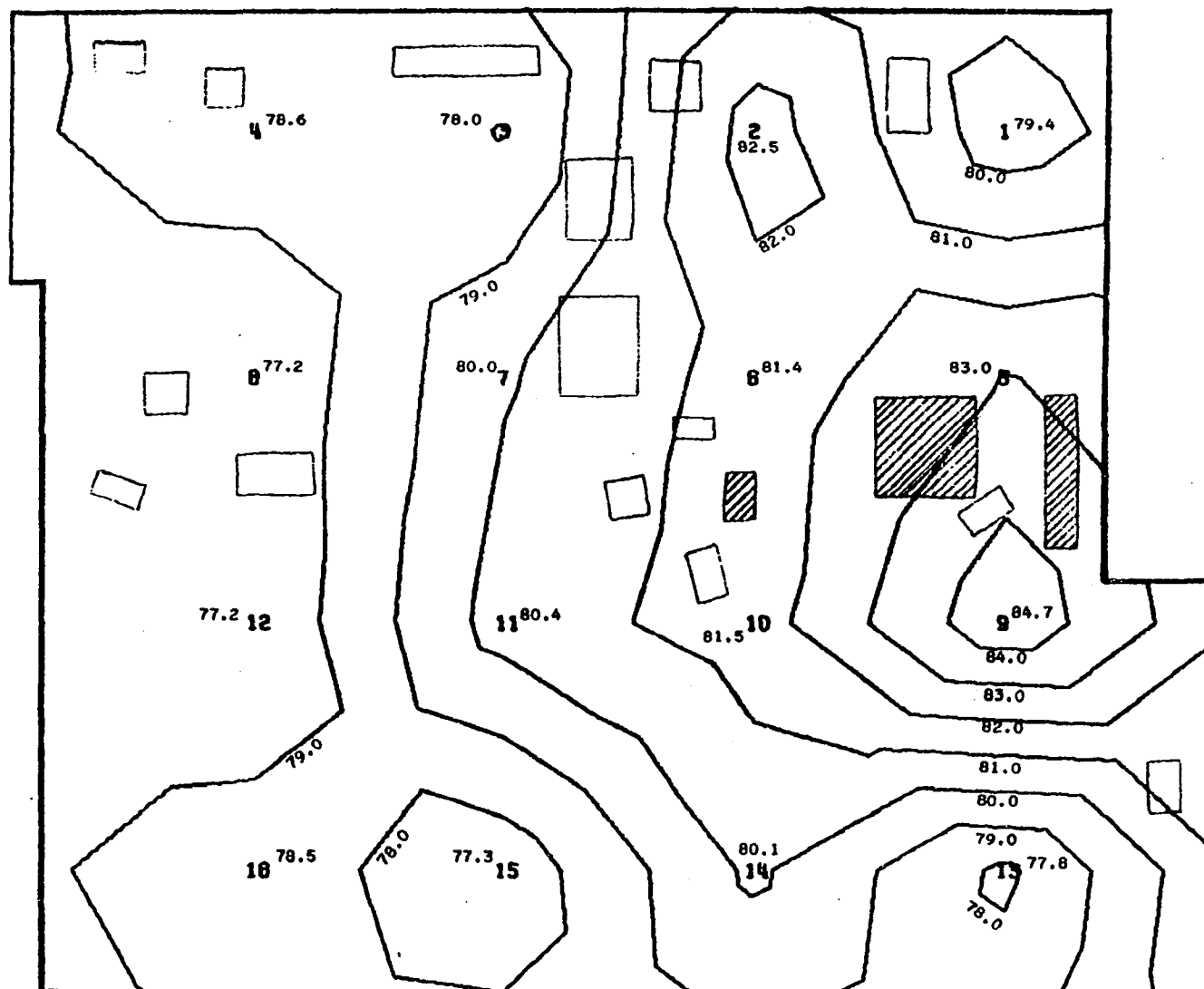


Figure 34. Treatment 13

34) produced a high level of sound in the entire laboratory, with the highest level being at microphone nine. In the case of treatments six and seven, the high readings were in the areas of microphones five and six. This pressure shift from west center to east center of the laboratory over the surfacer and jointer apparently resulted from the characteristic dominance of the uniplane around microphone nine. One point of interest is the addition of the readings of microphone nine for treatments six, seven, and eight in accordance with the chart for adding decibels (Figure 3). The sum of these three readings was 84.8 dB, which was within .1 dB of the microphone nine reading recorded during treatment 13.

Treatment 14 consisted of three small machines as shown in Figure 35. The point of maximum sound level centered around microphone 10, which was located near the dominant noise source of the group.

The contours of Figure 36 followed the general trend for single machines in regard to overall range throughout the laboratory. In this case the range was 11 dB, which produced 11 contour levels. Since the contours were generated at one dB intervals, the general range can be noted at a glance by observing the number of contour lines.

The contours of Figure 37, because of their number (14 levels), indicated the most extensive overall range of the treatments analyzed. The contours radiated in an even manner, which indicated a minimum of interaction during this particular treatment.

The table saw used for treatment 17 (Figure 38) produced a relatively simple set of contours as compared to the previous treatment. This saw, being a larger and more powerful machine, tended to fill the

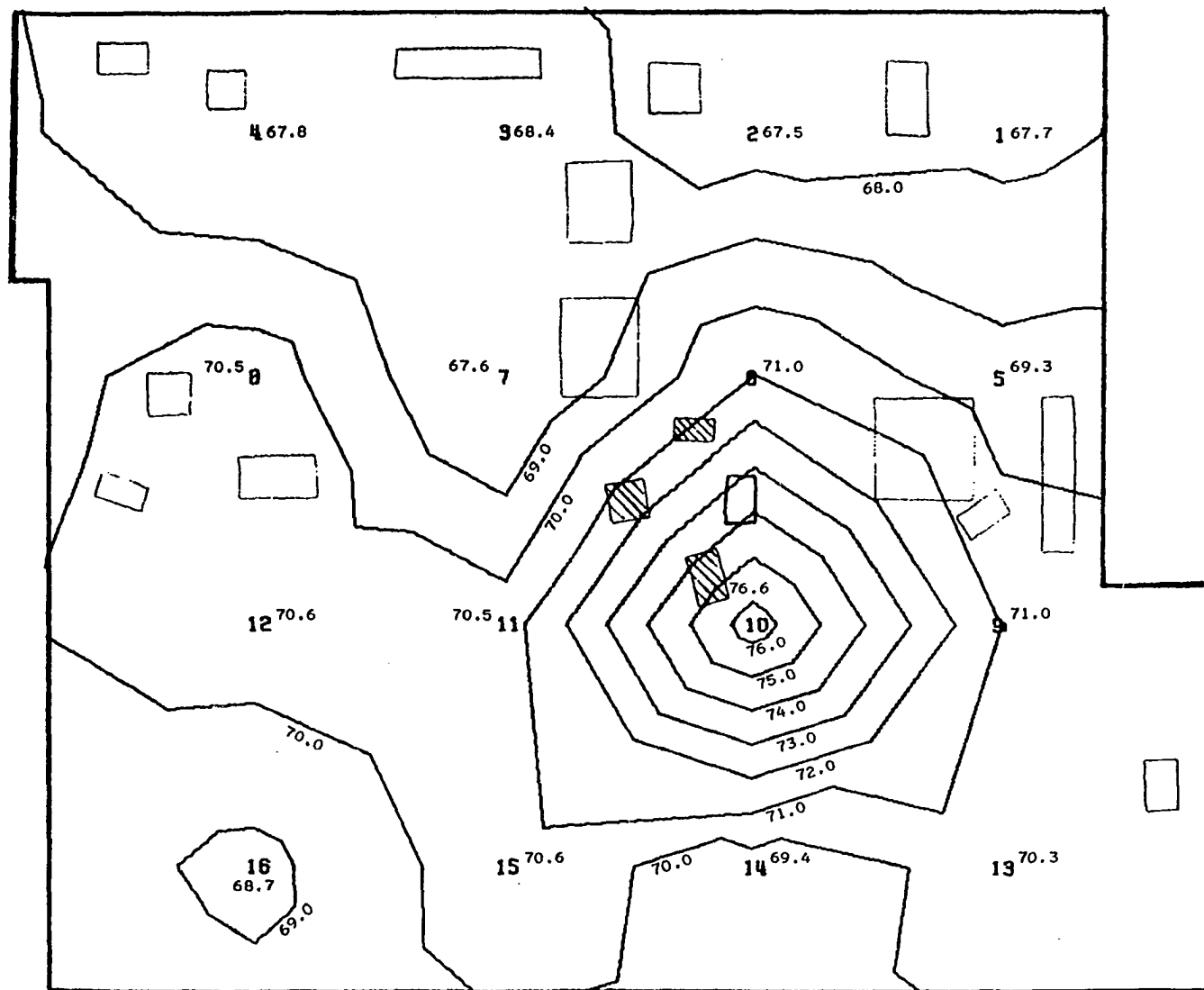


Figure 35. Treatment 14

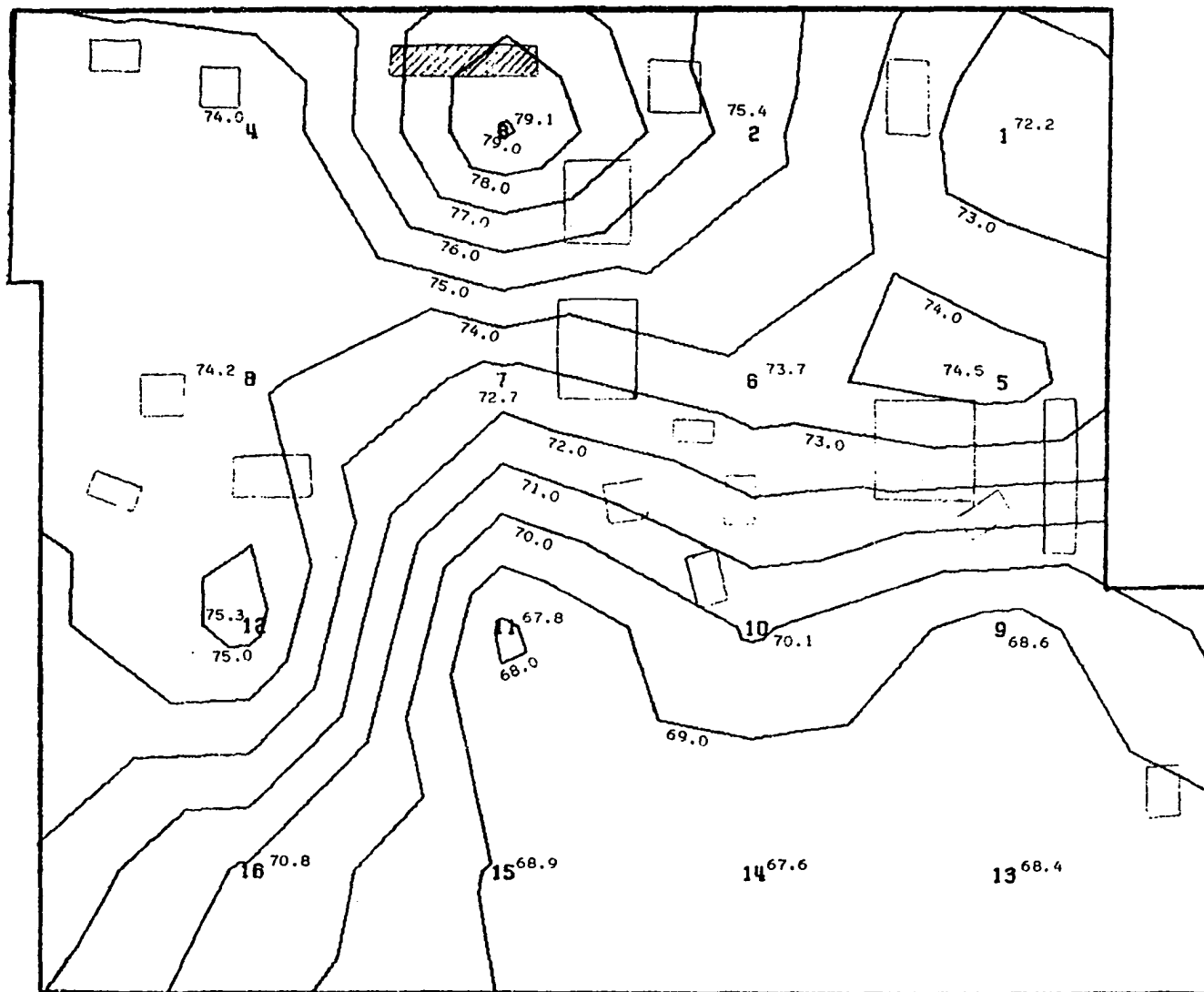


Figure 36. Treatment 15

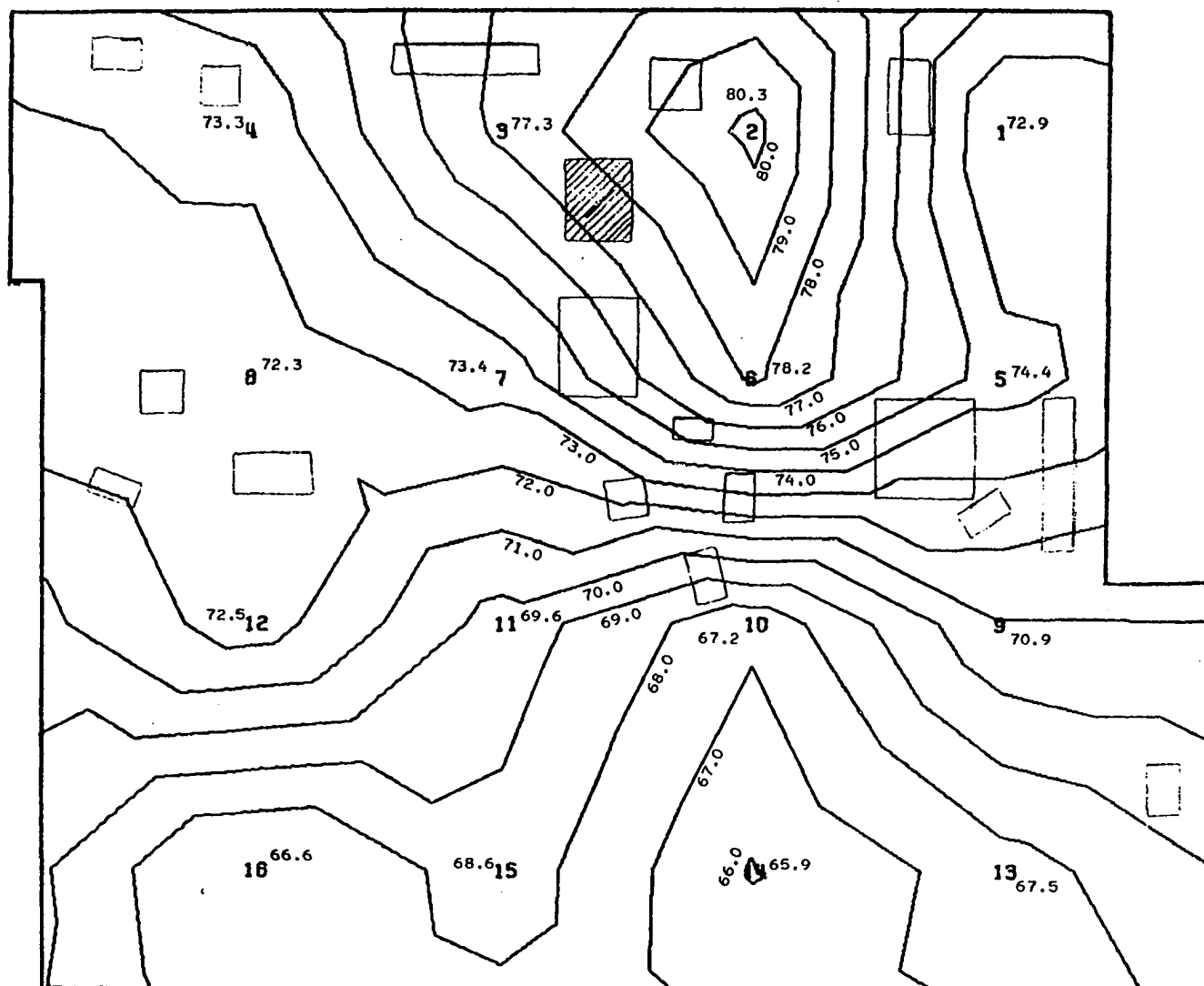


Figure 37. Treatment 16

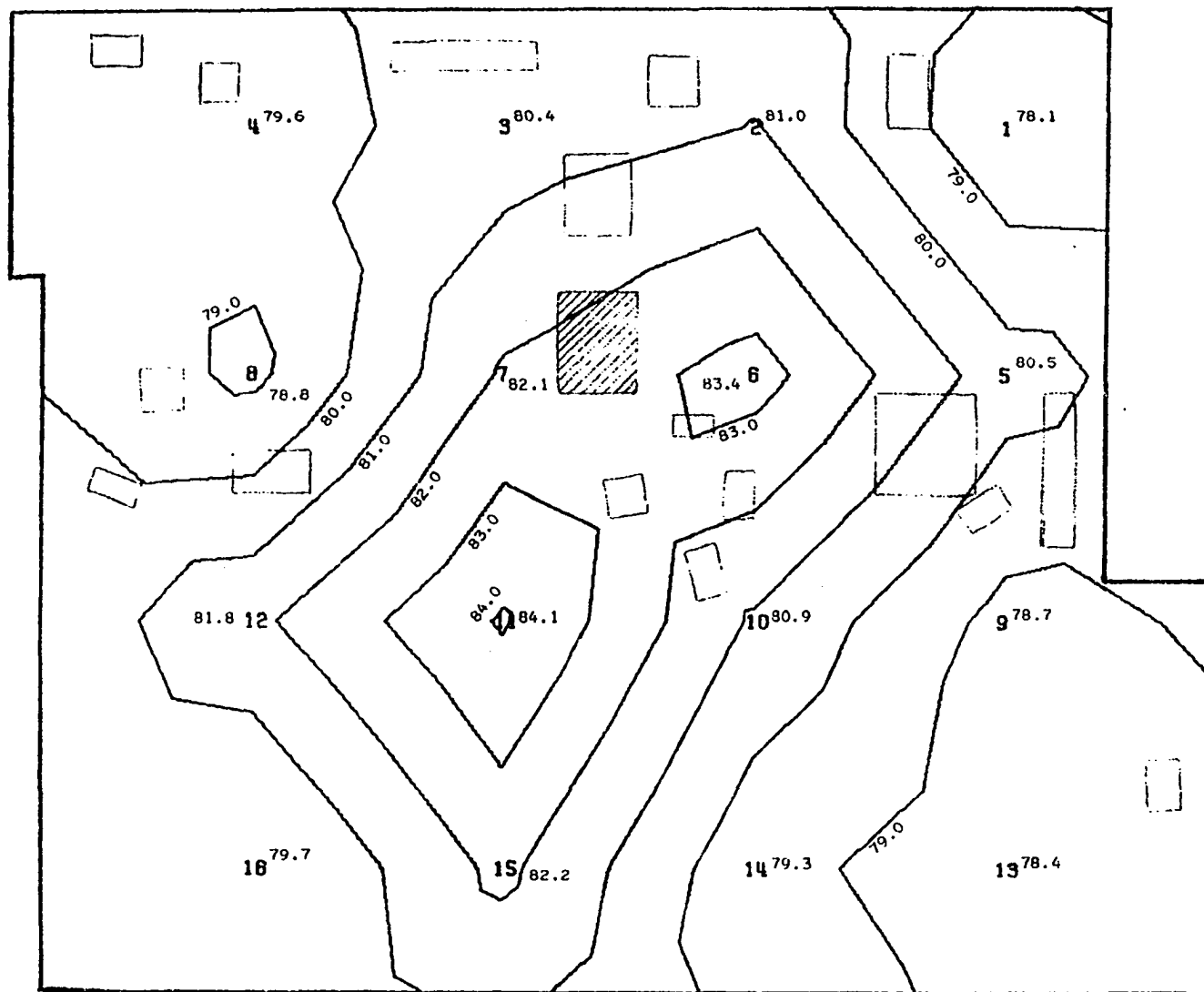


Figure 38. Treatment 17

laboratory with sound, which in turn reduced the overall range to five dB as indicated by the five contour levels.

For treatment 18 (Figure 39), a combination of machines used singly in treatments two, three, and four were used. When the method of decibel addition illustrated in the chart in Figure 3 was used, the microphone one readings added up to equal the value recorded on microphone one reading of treatment 18. The nine even contours show an overall range of nine dB.

The three machines of treatment 19, as shown in Figure 40, filled the laboratory with a relatively high level of sound. The high level of 85 dB surrounded the sound sources and radiated evenly to a low reading in the northeast corner of the laboratory. This treatment was a combination of treatments 15, 16, and 17, and by using the combining technique of Figure 3 and the reading of microphone eight, these treatments added up to 85 dB, which was equal to the treatment 19 reading for this microphone. Microphone eight was arbitrarily selected; however, the same technique applied to any of the 16 microphones gave consistent results.

The machines of treatment 20 (Figure 41) produced noise of sufficient intensity to cover the entire laboratory. The reduced number of contour levels (5) indicated the reduced overall sound level range.

All of the machines, with the exception of the heaters and the dust collector, were used for treatment 21 (Figure 42). The entire confines of the laboratory were subjected to a relatively high level of sound as indicated by the small number of contour lines.

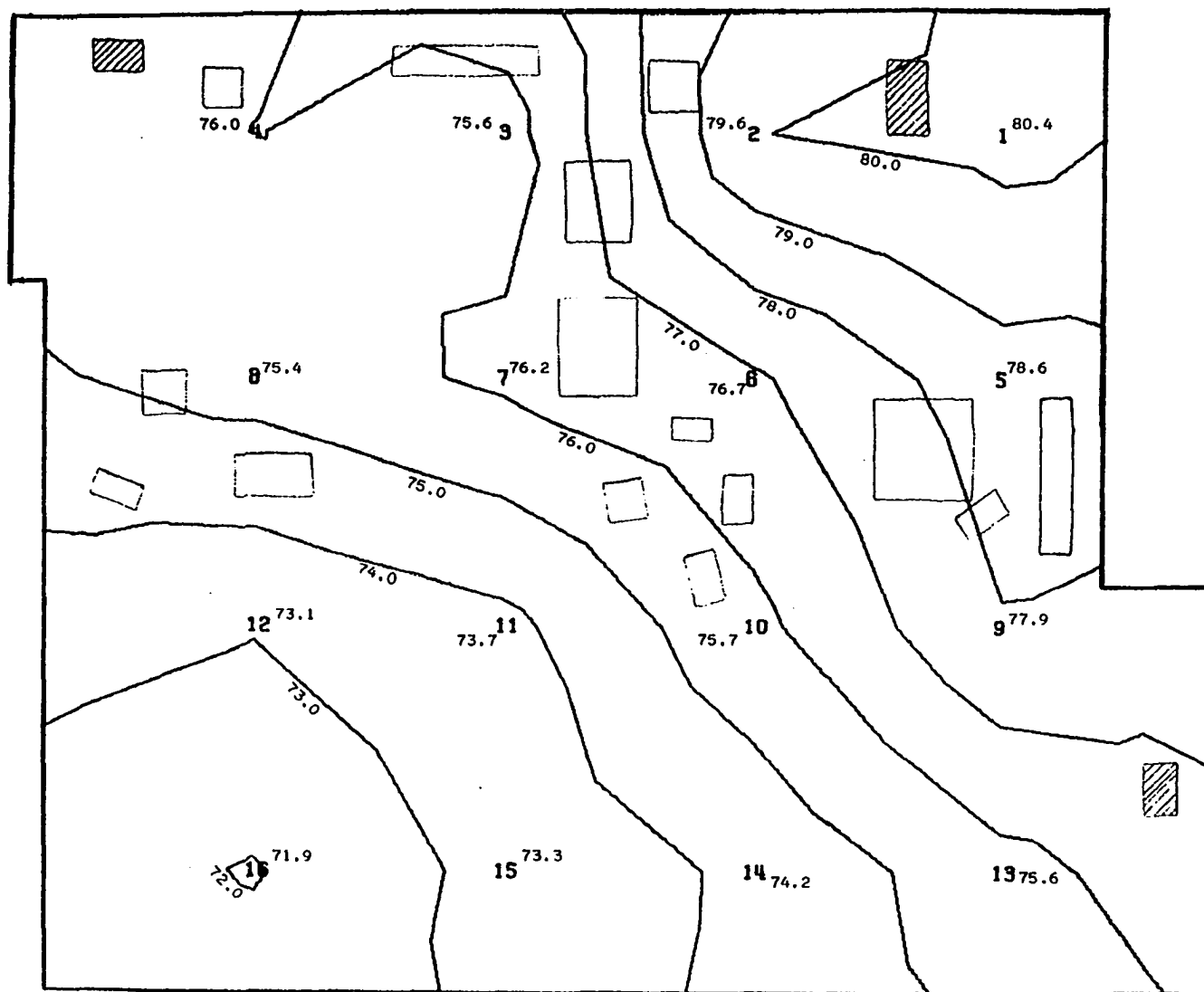


Figure 39. Treatment 18

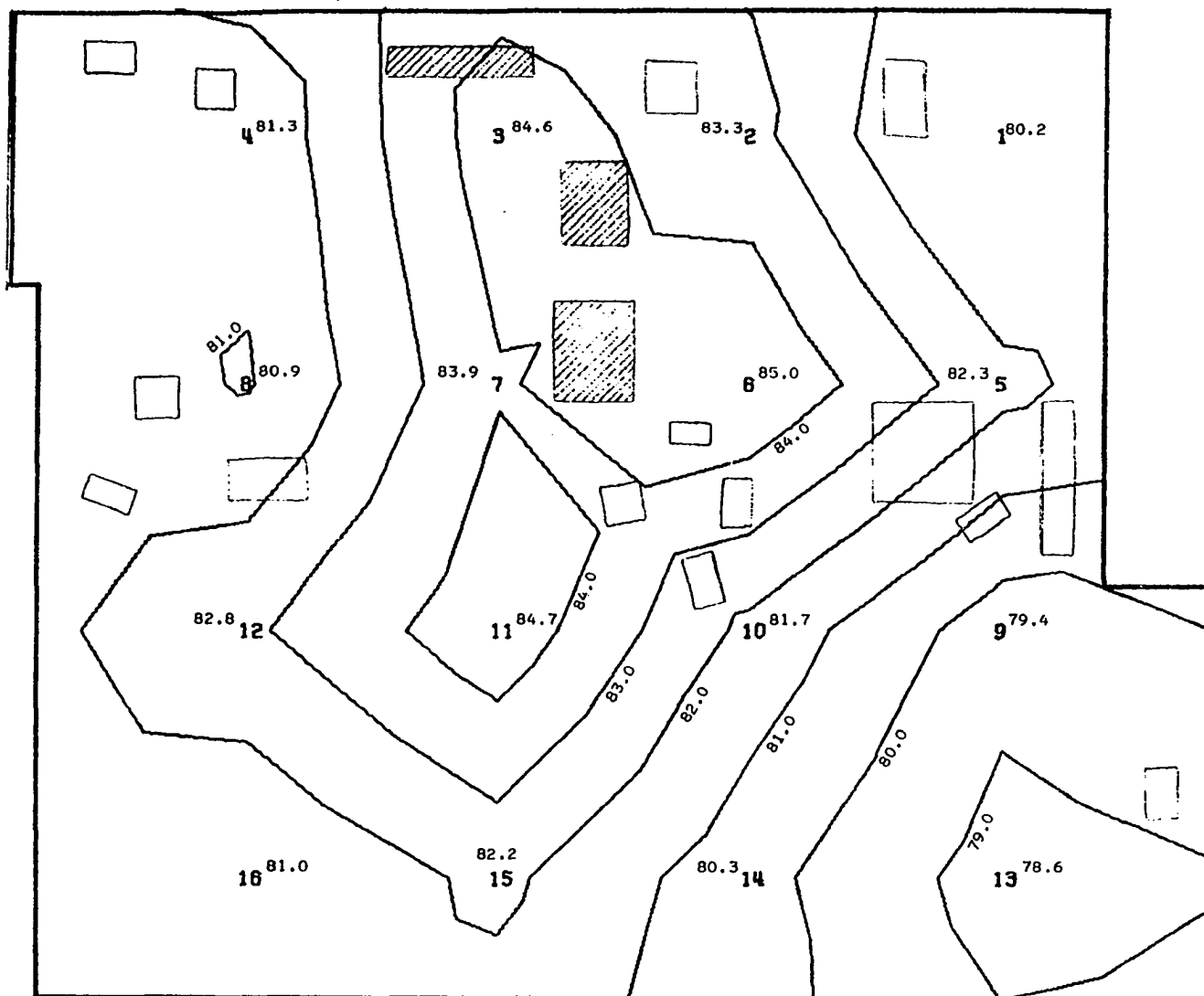


Figure 40. Treatment 19

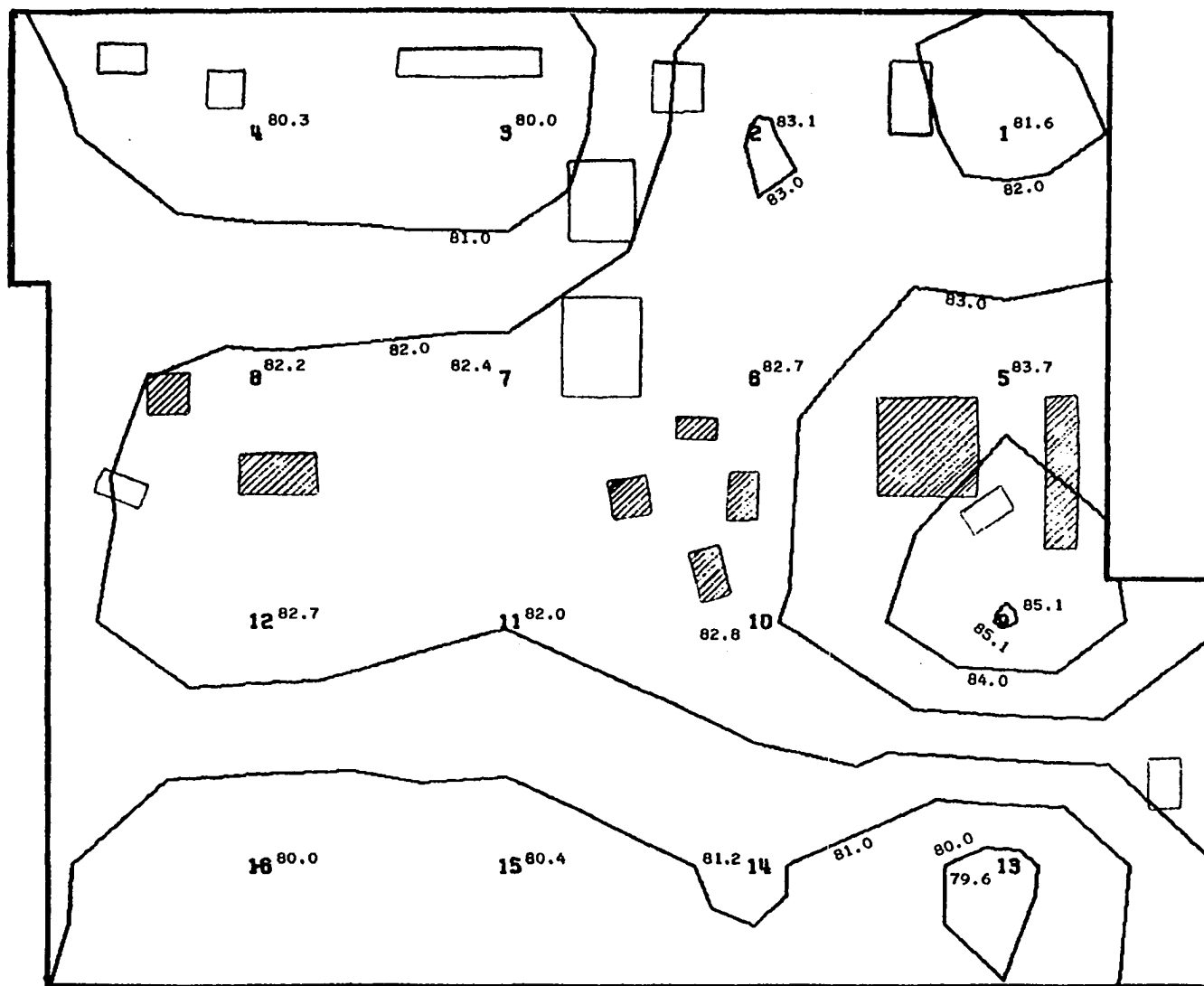


Figure 41. Treatment 20

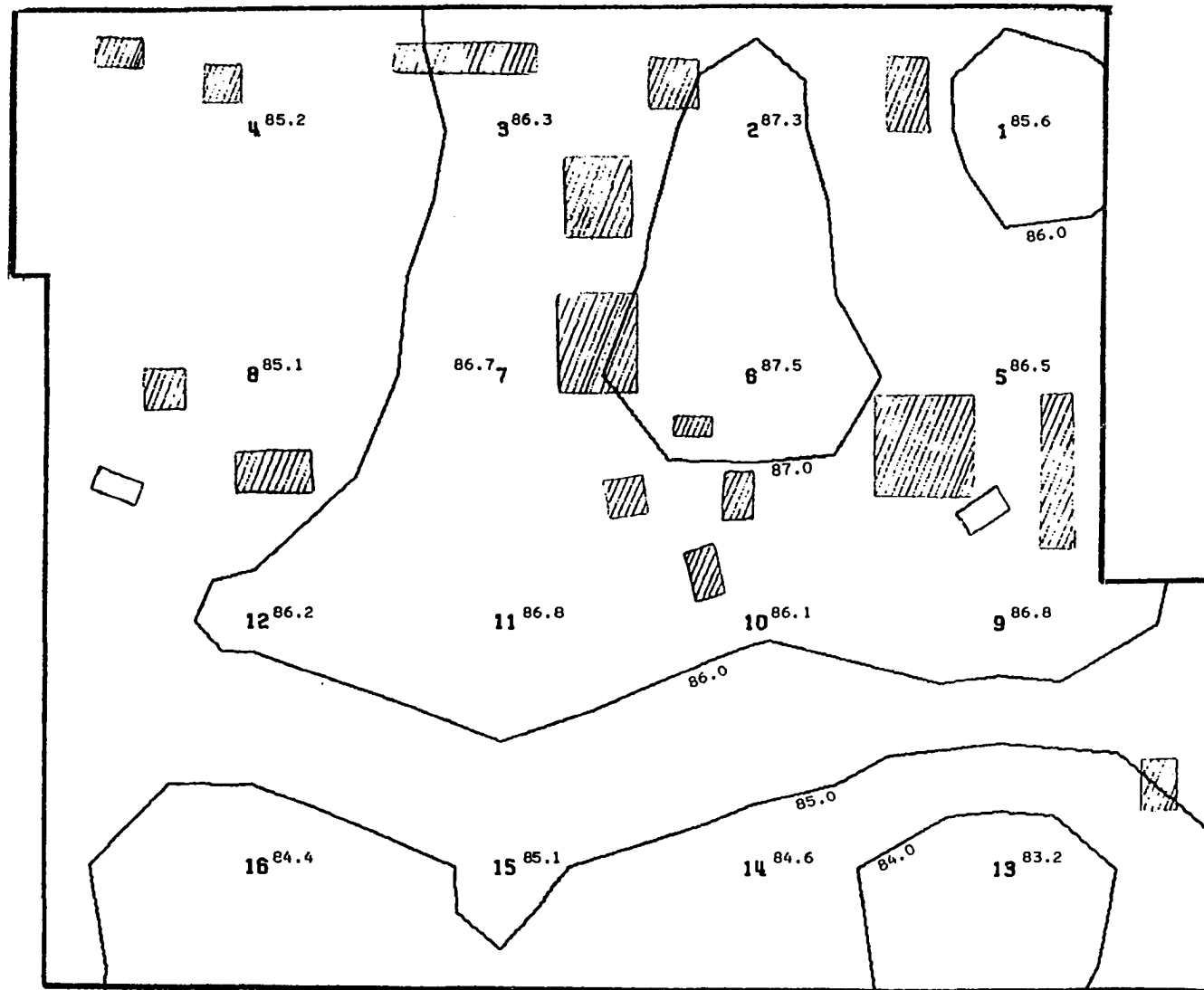


Figure 42. Treatment 21

The contours for treatment 22 (Figure 43) indicated that the addition of the dust collector increased the maximum level by two dB and added one contour line.

The final treatment in Figure 44 was almost unchanged from the previous treatment. This was expected since the level of the heaters which were added are 20 dB below the levels recorded during treatment 22. As can be noted from Figure 3, the addition of a sound source of this intensity caused no total change. This combining of decibel readings can be misleading; however, decibel levels are not added directly, but instead are converted to relative powers before being added or subtracted as the case may be (Peterson and Gross, 1974, p. 9). After adding or subtracting, the result is then converted to decibels. The chart in Figure 3 provides a simple method of performing this combining operation by stating the variables in terms of the difference in two levels. As can be noted from this chart, combining two equal levels adds three dB with the resultant total being 83 dB instead of 160 dB. When using this chart for combining unlike levels, the amount to be added is determined in terms of the difference in dB, and this amount added to the larger of the two. In the case of treatment 23, where the heaters were 20 dB below the level produced by the other machines, it can be noted that the scale on the chart ends at 15 dB of difference between two levels, at which point less than .2 dB is added to the greater reading.

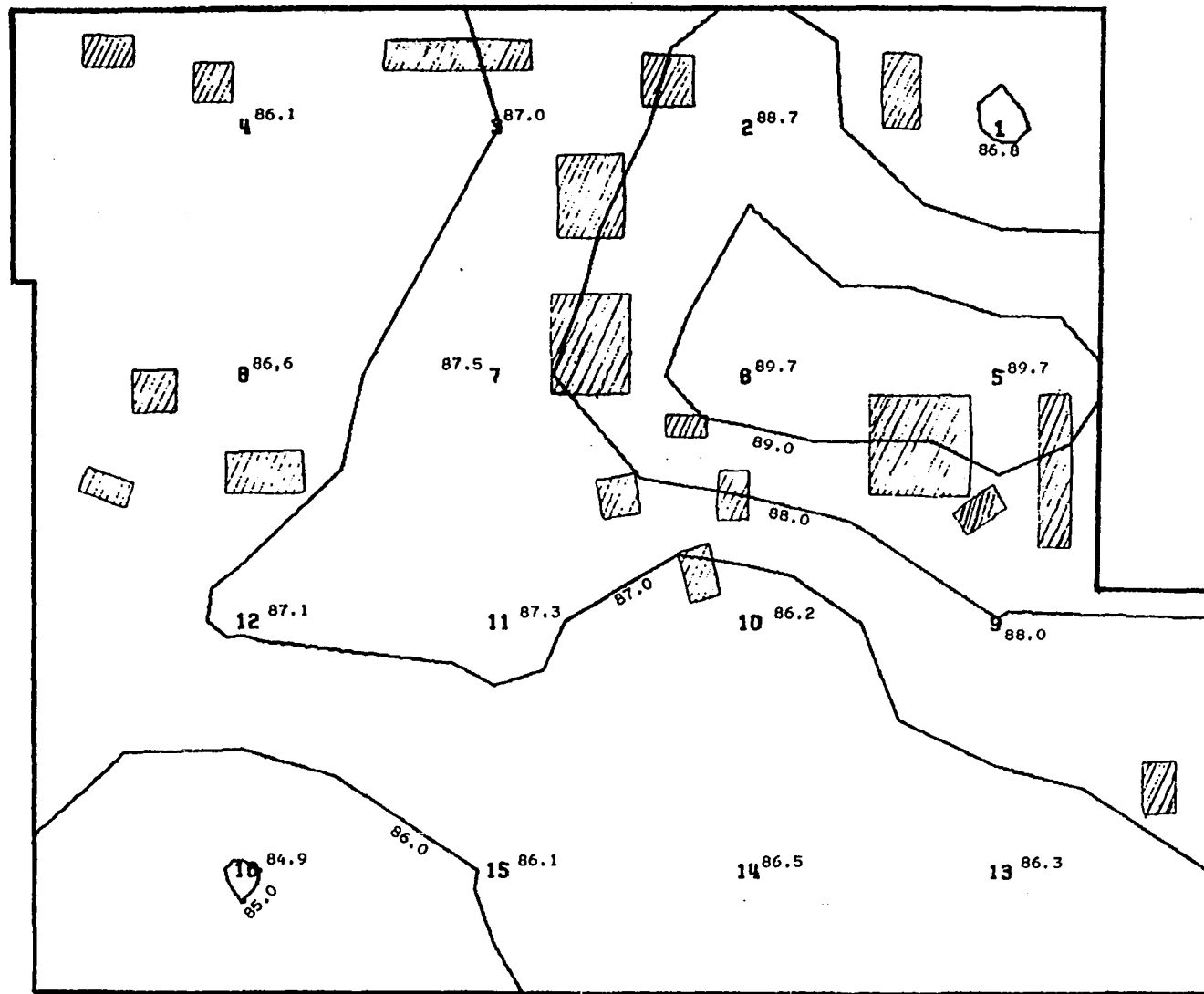


Figure 44. Treatment 23

CHAPTER V. FINDINGS

The objective of the study was to design and fabricate a sound monitoring system capable of sensing and recording for computer processing of data, multi-locational sound levels over extended periods of time. In regard to this objective, three questions were presented:

(1) Is the sound monitoring system responsive to locational changes of variable noise levels emanating from identified sources?

(2) Is the sound monitoring system capable of providing repeated recording of sound levels within acceptable tolerances?

(3) Is the system adaptable for detecting and recording noise levels in typical industrial arts laboratories?

It is important to reiterate that this study was not concerned with the design of a sound level measurement standard. An approved instrument was used as the standard. Within the instrumentation system, the only question of accuracy was concerned with the effect of the microphone switching system on the accuracy of the sound level meter (SLM). It was found during the initial calibration procedure that the SLM, when properly calibrated, responded with the same readings on its readout meter when used alone or with the constructed system. Also, within the system, the analog to digital conversion process proved to be accurate through prolonged tests with a standard calibrator. Repeated tests at various counts from 10 to 200 registered an error of \pm one count not in excess of five times per 100 readings. In normal use, the SLM readout meter cannot be visually read with greater accuracy.

The questions presented, which were concerned primarily with the external accuracy or compatibility of the system with the monitoring requirements, were as follows:

Question 1: Is the sound monitoring system responsive to locational changes of variable noise levels emanating from identified sources?

In the review of literature, the physical laws explaining the reduction in sound level as the distance from the source is increased were discussed. Although the wood laboratory, as a confined space, tends to augment or change this natural reduction with distance, it was apparent from the data readout in Figure 19 that the levels recorded increased or decreased as microphones in the matrix nearer or farther from the source were activated (refer to Figure 18 for laboratory layout). To better understand and compare the effects of multi-position sound sources, the computer-generated sound contour maps were developed. These contour maps served to converge the relatively disassociated information of the computer readout to a two-dimensional laboratory perspective which were easily interpreted and compared with contour maps for other sound sources or treatments. The contour maps in Figures 23, 24, and 25, where the treatment machines were located in different corners of the laboratory, clearly demonstrated a high degree of repeated responsiveness of the instrumentation to both level and locational changes.

Question 2: Is the sound monitoring system capable of providing repeated recordings of sound levels within acceptable tolerances?

It can be seen from the data sheet in Figure 19 that for 16 microphone positions and 23 treatments, with four matrix cycles per treatment,

the system was required to record 368 groups of four readings each. It was found that the average variation of the four readings was within \pm one dB, with the exception of the test item entered in the program for treatment 10, as discussed in Chapter IV. This variation was within the limits which would result from changes in background noise level, periodic variations in the noise source, and sound wave interaction with the laboratory enclosure. These repeated readings established the accuracy or repeatability within the instrumentation system over a monitoring period of several hours.

Another point of interest noted during the course of this study was that after the system was idle for periods of six weeks, it again recorded readings accurate within one count without any calibration or warm-up period. This stability can be expected from solid state circuitry operating from well-regulated voltage sources.

Another facet of the question of repeatability was from the standpoint of session-to-session repeatability. Since this system automatically monitors from a predetermined fixed program, subsequent monitoring sessions could be made without the introduction of human variables introduced by hand-held instrument methods. The contour maps provide a permanent record of data for comparison of treatment effects with the original sound environment.

Question 3: Is the system adaptable for detecting and recording noise levels in typical industrial arts laboratories?

This study was basically concerned with A-weighted sound levels in the OSHA range from 85 to 115 dB. As was explained in Chapter III, the

AD converter in this system produced accurate counts from 20 dB below the center scale reading of the SLM to 7.6 dB above. The SLM has a series of 10 dB ranges from 50 dB to 130 dB plus a special OSHA range of 85 dB to 115 dB. The center reading on the SLM is the range setting plus four dB, with the total measurement capability for a given range being 27.6 dB. In this study where the sound levels were not in excess of 90 dB, the 80 dB range on the SLM was used, which provided a satisfactory range from 64 dB to 91.6 dB. For OSHA purposes in school laboratories, the 84 dB to 111.6 dB capabilities of the 100 dB SLM range would generally be adequate. If, however, higher levels were encountered, the special OSHA range with a center reference reading of 109 dB could be used. It is important to consider that since the system monitors automatically, the range selected must accommodate the sound level extremes encountered. Careful attention to overcount at the beginning of the session for this study made it possible to select the correct range. By proper selection of appropriate range, any sound environment within OSHA specifications can be recorded.

In summary, the question can be asked: What are the advantages of this system as compared to a hand-held sound level meter? Two important factors will be considered: (1) the capability to collect and process data, and (2) measurement repeatability.

In a five-hour monitoring session with readings every five seconds, there would be 225 matrix cycles or 3600 readings. These data would occupy 30 feet of punched tape which can be read into the computer in a few seconds. These data can be computer processed and read out in any

desired form. The capabilities to accomplish this through manual means would not be possible.

A valuable facet of sound measurement rests with the determination of the effectiveness of corrective measures that have been made in a given situation. Since this system can be installed and programmed in subsequent monitoring sessions in a manner identical to the original or pretreatment session, the repeatability of the measurements would be free from human variables.

CHAPTER VI. CONCLUSIONS AND RECOMMENDATIONS

Conclusions

In fulfillment of the purpose of this study, a sound monitoring system was developed to sense and record for computer processing at pre-determined intervals, the sound levels present at varied locations in industrial arts laboratories.

Specifically, the system was found to be responsive to locational changes of variable sound sources. This was apparent in a broad sense from the row-column data readout, but when this information was converted by computer processing to sound contour maps, a visual record of the laboratory sound profile for a particular treatment was created.

This instrumentation system was found to be capable of providing repeatable readings of various sound situations. Under laboratory conditions with an approved calibrated sound source, the device was consistent within $\pm .1$ dB mid-scale. For this study, with varied machines used at treatment sound sources, the four readings for each microphone in the 16 microphone matrix were consistent with one another within the bounds of the acoustical variables in this laboratory.

Finally, it was found that the instrumentation system possessed range capabilities to monitor any sound within the full spectrum that may be encountered in industrial arts laboratories.

Recommendations

Based on the performance characteristics this instrumentation system was found to possess, the following recommendations are made:

- (1) This system should be used for sound research in a laboratory using different equipment and facility design.
- (2) This system, because of precise sampling capabilities in the time domain, could be useful in correlated sound and performance studies.
- (3) This equipment can be adapted to record on command from one hand-held microphone, thus providing automatic recording of manual measurements.
- (4) Further research with microphone placement and matrix arrangement should be considered.
- (5) Research in the development of computer programs specifically for sound contour maps and the treatment of the readout data should be initiated.

Limitations

- (1) The sophistication of computer programs for processing sound measurements is still in the early developmental stages.
- (2) Further extension of microphone cables necessitates the use of microphone preamplifiers.

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ACKNOWLEDGEMENTS

It is with the assistance and encouragement of many, many people that an endeavor of this type can be completed. I particularly want to recognize the assistance and direction given by my major professor, Dr. William Wolansky, and the other members of my committee, Dr. Ross Engel, Dr. Robert Gelina, Dr. Gerald Parks, and Professor Jack Shelley.

To the REAP staff, thank you for your help with test instruments and equipment.

I appreciate the cooperation and interest of the Simpson Electric Company in the study.

To my wife, Verna, who gave me her undivided support.

APPENDIX:
SCHEMATIC DIAGRAMS

